VOL. 9
NO. 28

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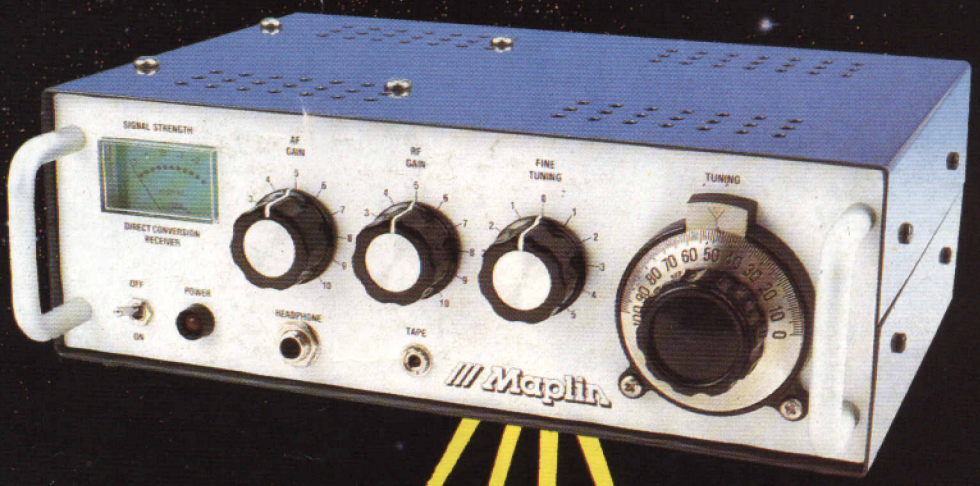
Electronics

THE MAPLIN MAGAZINE

TUNE-IN TO THE WORLD WITH THIS SUPERB SHORTWAVE RECEIVER



Also in this issue:

LOUDSPEAKERS
The first 111 years!

ELECTRONICS
by Experiment!

CALCS
for Engineers!

Plus
Speaker Cabinet Design, MW/LW radio, Stripboard Projects!

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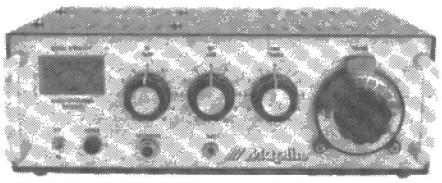

A choice of four tuning kits allow you to listen to speech or morse code from 10 metres to 160 metres. The main features of this easy-to-build unit are a tuned active front end, buffered RF oscillator output, reduction drive 'vernier' tuning and on-board voltage regulator and audio power amplifier.

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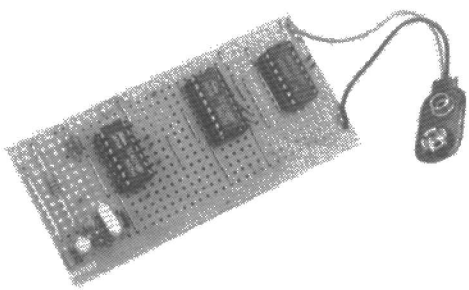
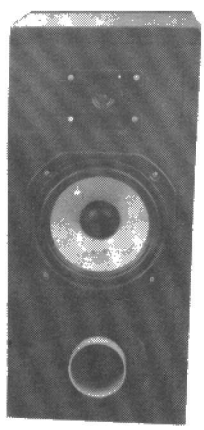


Four useful mini circuits from the prolific pen of Robert Penfold, built on stripboard. You can construct a MIDI thro box, a capacitance meter adaptor, a crystal calibrator and a transistor tester.

2 Way 16 Litre Loudspeaker System 56

Another in our popular series on loudspeaker cabinet designs. Full cabinet construction details are given and the electronics and baffle are available.



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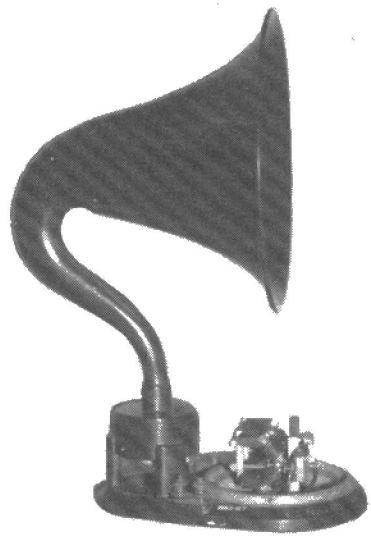

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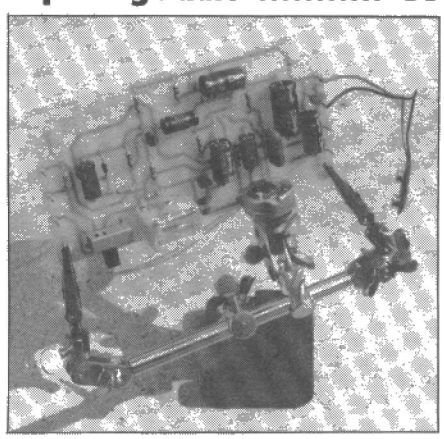

The MW radio in the last issue is expanded to cover the LW band also. A printed circuit board is available so that you can build it for yourself.

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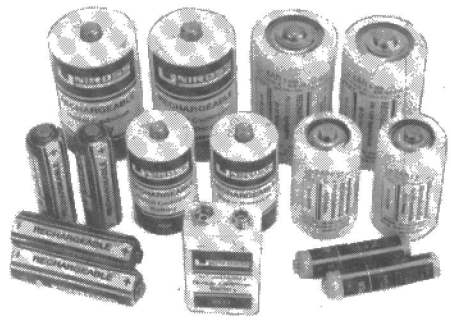

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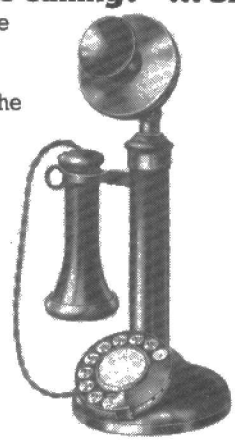

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DIRECT CONVERSION SHORT WAVE RECEIVER

by Chris Barlow

- ★ Receives Speech (SSB, DSB, AM) and Morse (CW)
- ★ Choice of Amateur band, 160 metre – 10 metre
- ★ Reduction drive 'Vernier' tuning
- ★ Tuned Active front end
- ★ Signal strength meter
- ★ Buffered RF Oscillator output
- ★ On-board Voltage regulator and Audio power amplifier

Specification of Prototype

RF Specifications (80 Metre version)

Tuning Range:	3.490MHz – 3.810MHz
Frequency Stability:	Less than 100Hz/hour drift after 30 minute warm-up
Sensitivity:	0.1μV for a readable signal 0.3μV or less for 10dB (S+N)/N
Oscillator Output:	450mV RMS off load 100mV RMS into 50Ω load

AF Specifications

Bandpass:	2.8kHz at –6dB (175Hz – 3kHz)
Dynamic range:	90dB
Signal to Noise:	40dB
Power Output into 8Ω:	1W RMS
Distortion:	1%
Tape Output:	100mV RMS into 47kΩ
Headphone Output:	8Ω – 32Ω Mono/Stereo

DC Specifications

Unregulated Power Input:	15 – 20V
Regulated Power Input:	10.5 – 14V
Quiescent Current at 12V:	147mA
Current at full Output:	295mA

Introduction

The receiver described in this article has been designed to have many of the features found on sophisticated ready-made short wave radios. The cost of such radio equipment is generally quite high and this can discourage newcomers to the hobby. However, good results can be achieved using relatively simple home-constructed receivers of the direct conversion design. This type of receiver has the advantage of simplicity of construction and ease of alignment, with the minimum of test gear.

The frequency range of the short wave bands start as low as 1.7MHz and extends up to 30MHz. To include all these bands on one receiver would present switching and tracking problems that would result in a compromise in its performance. For this reason it was decided that the receiver would cover only one band, but which one? Within the short wave spectrum there are segments allocated to amateur radio operators. These are people located all over the world pursuing the hobby of long distance communication using privately owned radio equipment.

There are six main amateur bands, with the addition of three new ones, and the choice of band is entirely up to you. However, the 80 metre band, admittedly not the best band for long distance reception will provide European stations after dark and British amateurs during daylight hours. When conditions are favour-

able and a suitable aerial is used, stations from further afield can be received, such as America, Africa and Australia. A popular higher frequency band is 20 metres, however this and even higher bands are affected by changes in world wide environmental conditions of the upper atmosphere. This tends to leave them inactive or 'dead' for much of the time and this fact should be taken into consideration when choosing the tuning pack associated with this project. A complete list of the HF amateur bands showing the relevant tuning pack you should use is shown in Table 1. DON'T FORGET TO ORDER THE TUNING PACK WHEN ORDERING YOUR RECEIVER KIT.

Direct Conversion

A direct conversion receiver achieves in one signal conversion operation what a superhet achieves in two or more. This is done by mixing the incoming RF signal in a non-linear device with a locally generated RF carrier close to the frequency of the incoming signal. One of the resulting products is the audio modulating frequency when receiving single sideband (SSB) or a beat frequency when receiving morse code (CW). This audio signal is then filtered out from the other unwanted mixer products and amplified, forming the audio output of the receiver.

Circuit Description

In addition to the circuit shown in Figure 2, a block diagram is detailed in Figure 1. This should assist you when following the circuit description or fault finding in the completed unit.

The receiver has two DC power inputs, regulated and unregulated. If using the regulated input, the voltage must be between 10.5 to 14V from an external DC regulated supply or batteries. When using the unregulated input, a mains adaptor such as the unregulated 1A (YM85G), set to its 12V output, will supply the necessary voltage to the regulator RG1. This will be in the order of 15 to 20V and RG1 will stabilise this voltage to 12V. It is most important that the DC supply be connected in the correct polarity, with the positive (+V) going to the centre pin of the DC connector SK1 or SK2.

When the power switch S1 is turned on the supply is connected to the main de-coupling capacitor C3 and the front panel indicator LD1. LP1 is a wire-ended filament bulb mounted behind the signal strength meter to provide back illumination. The +12V supply then feeds a second voltage regulator, RG2, which is a variable output type and is set to produce +8V by the resistors R2, 4, 5 and RV1. The preset RV1 is a 22-turn cermet type, which is used when setting up the highly accurate +8V varicap tuning reference. R3, C6 and C7 provide yet more supply de-coupling for the rest of the circuit. For the op-amps to function correctly a half supply reference must be generated, this is provided by one half of IC2. The voltage reference applied to the input of this op-amp is derived from the two resistors R6 and R7 which form a potential divider. The op-amp is merely used as a zero gain buffer to provide a low impedance half supply, its output being de-coupled by C80.

The aerial and earth from SK5 connects to pins P10,11 on the circuit board. To reduce the amount of 'out of band' signals reaching the RF amplifier, a series tuned circuit comprising of T1 and C10 is used. At its resonant frequency this circuit has a low impedance and will allow the RF to pass into the low impedance winding of T2. However, to all other frequencies this circuit appears as a high impedance, thus reducing the level of unwanted RF energy reaching T2. The output of T2 is a parallel tuned circuit, with C12, TC1 and the varicap diode VC1 controlling its resonant frequency. The RF signals across this circuit are then applied to the high impedance 'gate one' input of TR1. The other gate has a variable DC bias applied to it, derived from RV2, R19 and is de-coupled to RF by C14. As the bias voltage is increased,

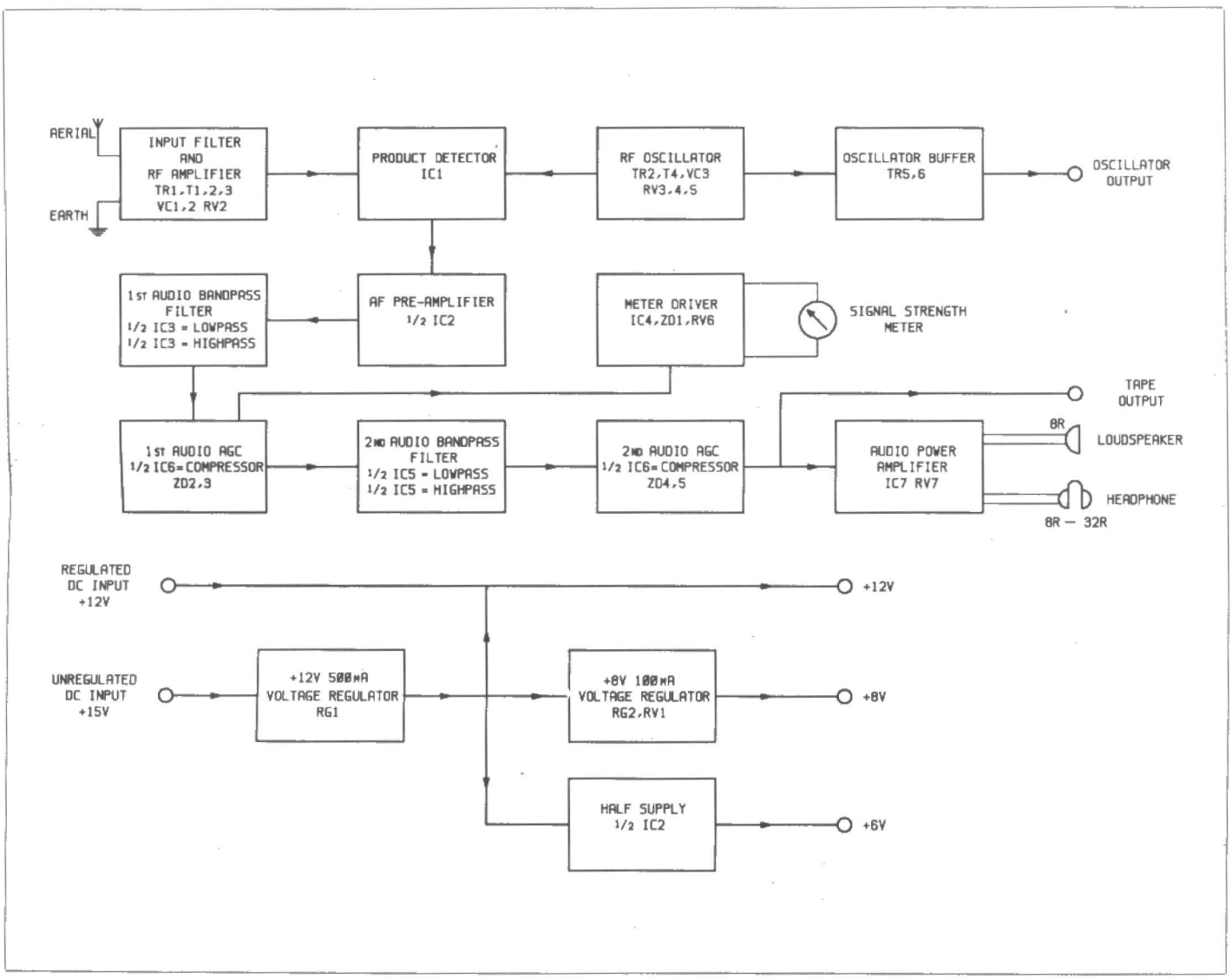

Figure 1. Block diagram.

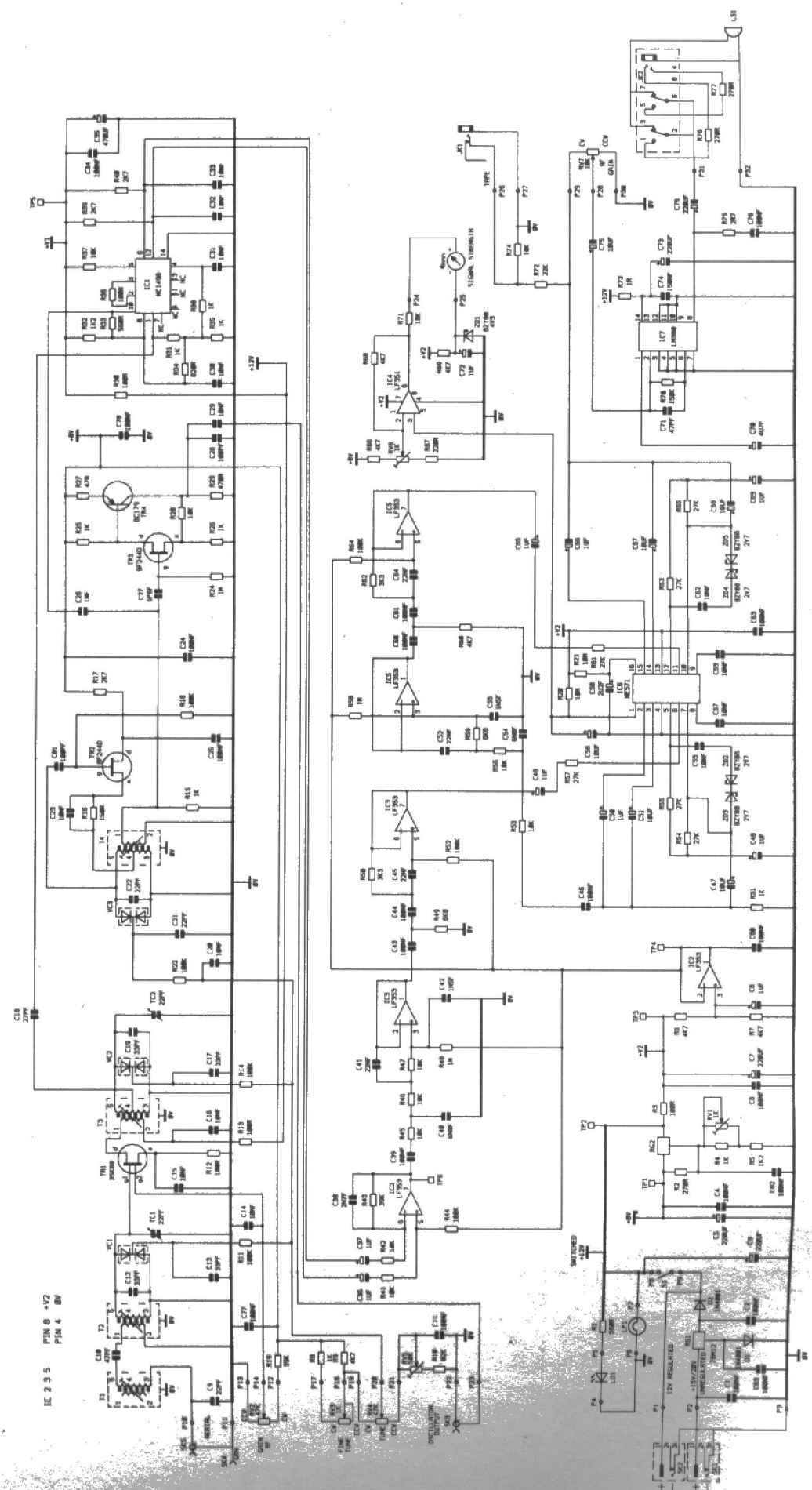

Figure 2. Circuit.

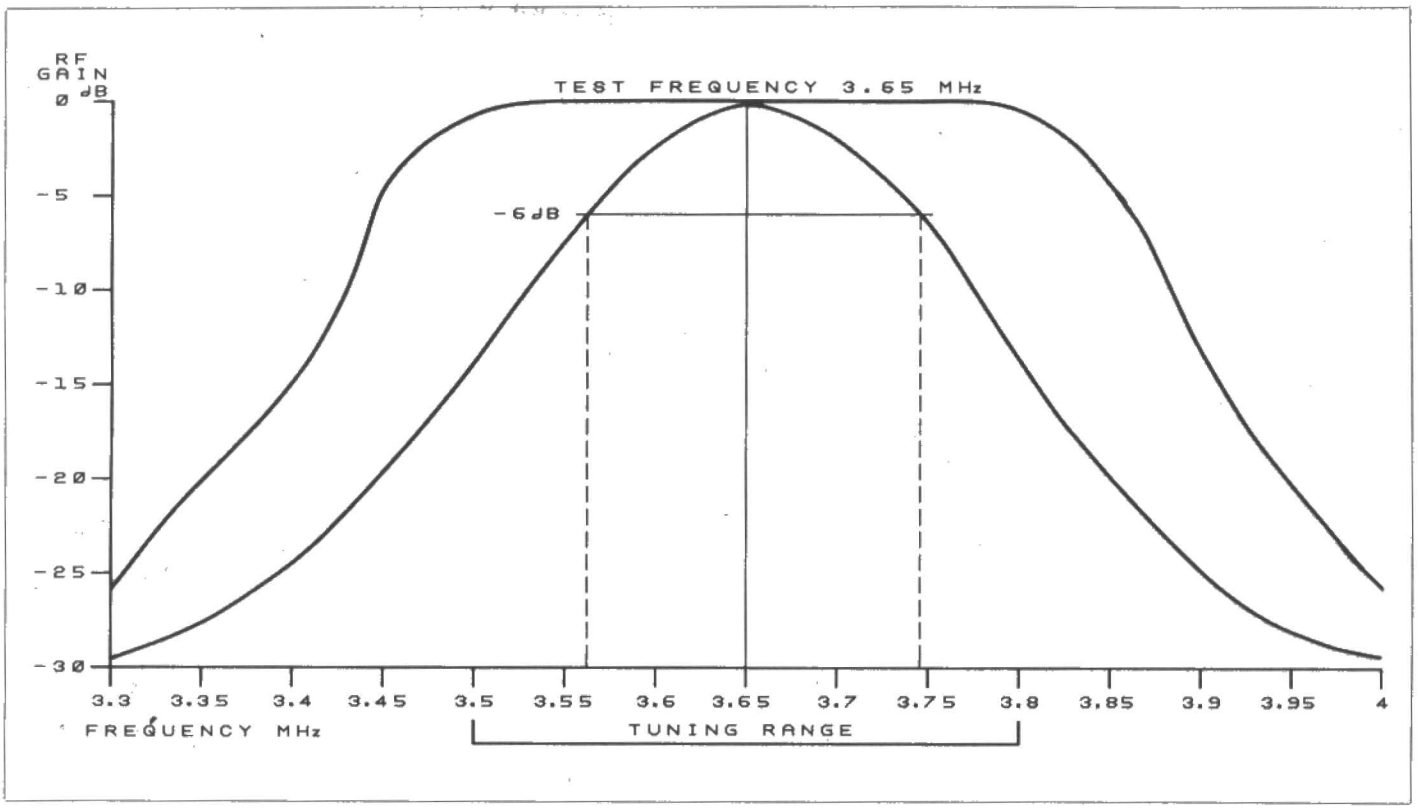

Figure 3. RF filter and amplifier response (80 metre version).

the gain of the amplifier will increase to provide almost 20dB of gain. The low impedance winding of T3 is placed in the drain of TR1 and its output is tuned by C19, TC2 and VC2. The RF signals are then taped off and fed via C18 to the product detector IC1. A graph showing the tuning and bandpass response of the prototype 80 metre front end is illustrated in Figure 3.

The RF oscillator consists of an FET transistor TR2 and T4 being tuned by C22, VC3, in the Hartley configuration. The output is taken from the low impedance winding of T4 and is fed to the product detector via C26. The output is also taken via C27 to a broad band buffer, comprising of TR3 and TR4. This provides a low impedance output for driving frequency counters or other devices. The DC bias used to tune the oscillator and RF amplifier is generated by a chain of resistors. The high end is set by the value of R8 and fine tuning is provided by RV3. RV4 is used for the main tune control, with the 22-turn preset RV5 and R10 setting the low frequency limit. A plot of the frequency stability of the prototype 80 metre oscillator is shown in Figure 4.

An MC1496 double balanced mixer IC1 is used to produce the sums and differences of the two frequencies applied to pins 1 and 10. The audio output we want is the difference frequency between the RF signals from the input amplifier and the output of the RF oscillator. IC1 produces two audio outputs which are in anti-phase to each other. These signals are fed via C36, C37 to the inverting and non-inverting inputs of IC2. This op-amp is used to produce a high level signal for the bandpass and compressor circuits.

The audio processor comprises of four stages, two bandpass filters and two compressor circuits. Each bandpass filter is made from a lowpass and highpass circuit using a dual op-amp, the first being IC3 and the second IC5. The combined effect of these

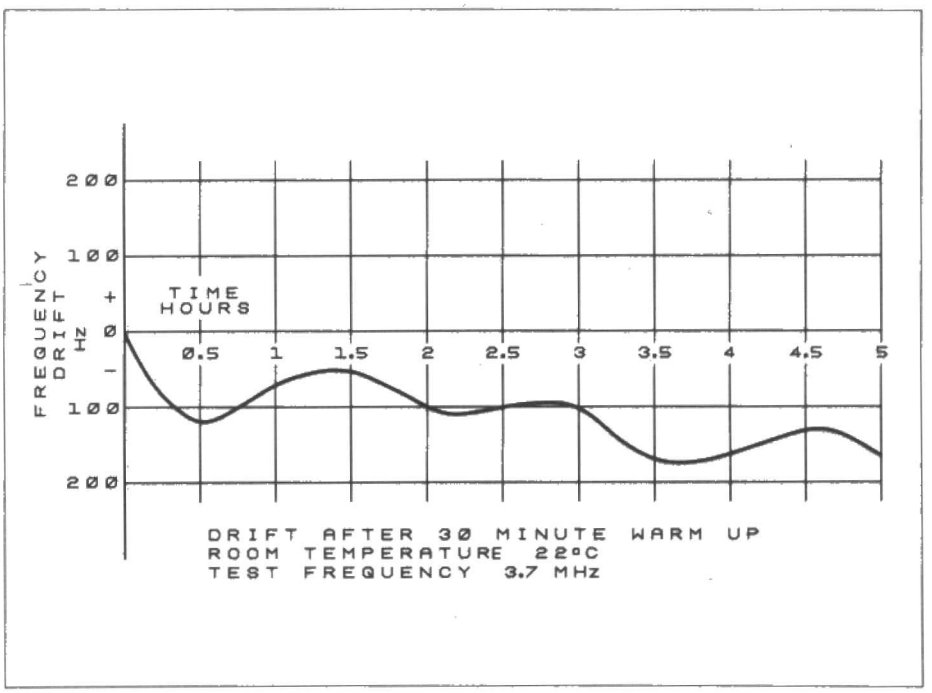

Figure 4. Frequency stability.

filters can be seen in Figure 5. A compressor is used to provide an automatic gain control (AGC) to reduce the volume differences between strong and weak stations. The DC reference produced by the first AGC stage is used by the meter driver IC4 to indicate the strength of the signal. RV6 on pin 2 is used to set the meter zero point and as the voltage on pin 3 progressively increases from the AGC stage the meter will show a steadily rising value. The combined affect of the two compressors is shown in Figure 6 and the signal strength meter response in Figure 7.

The output from the second compressor feeds the final audio product to the tape output and the AF gain control RV7. The sound output stage IC7, the LM380, is capable of driving an 8Ω loudspeaker or 8 to 32Ω headphones. Stereo or mono phones can be used and when plugged into the quarter inch jack socket JK2 the speaker is automatically switched out.

PCB Assembly

The PCB is a double-sided, plated-through hole type, chosen for maximum reliability and stability. However, removal of a misplaced component is quite difficult with this kind of board so please double-check each component type, value and its polarity where appropriate, before soldering! The PCB has a printed legend to assist you in correctly

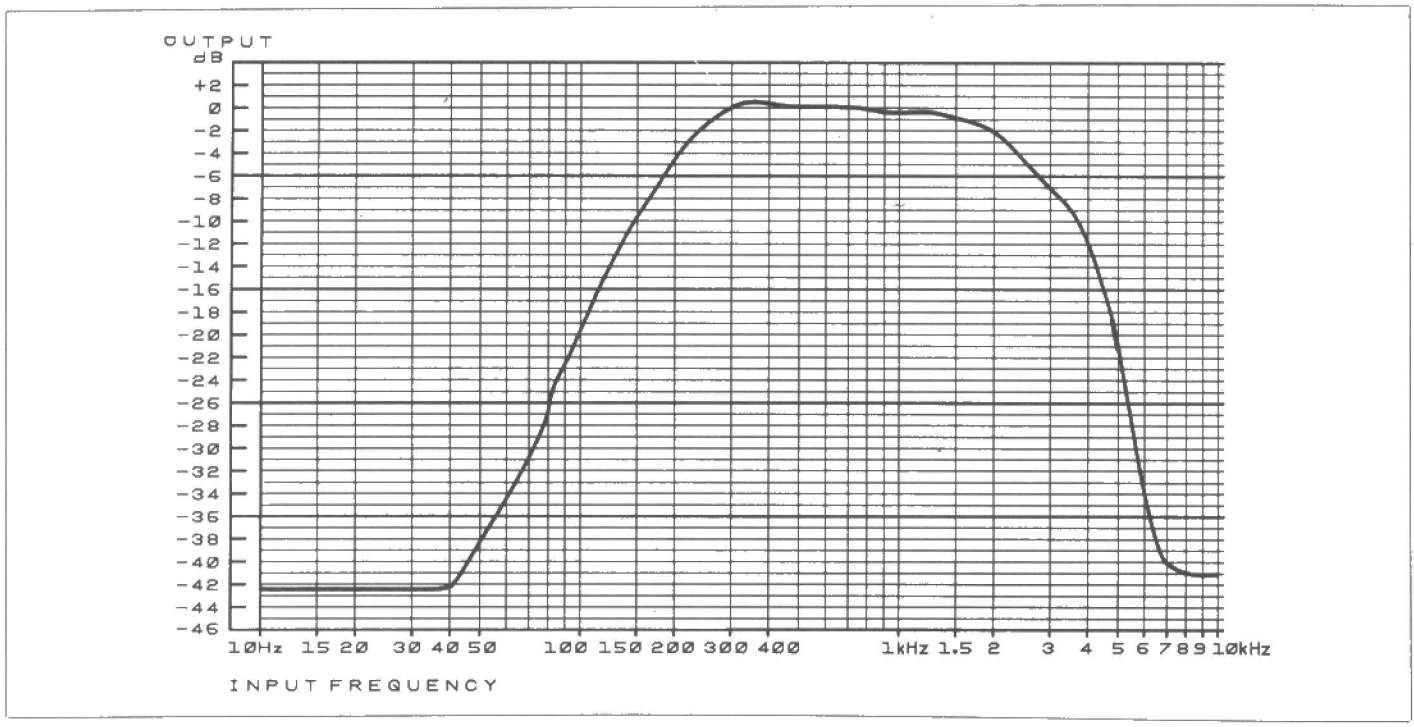

Figure 5. Audio bandpass filter response.

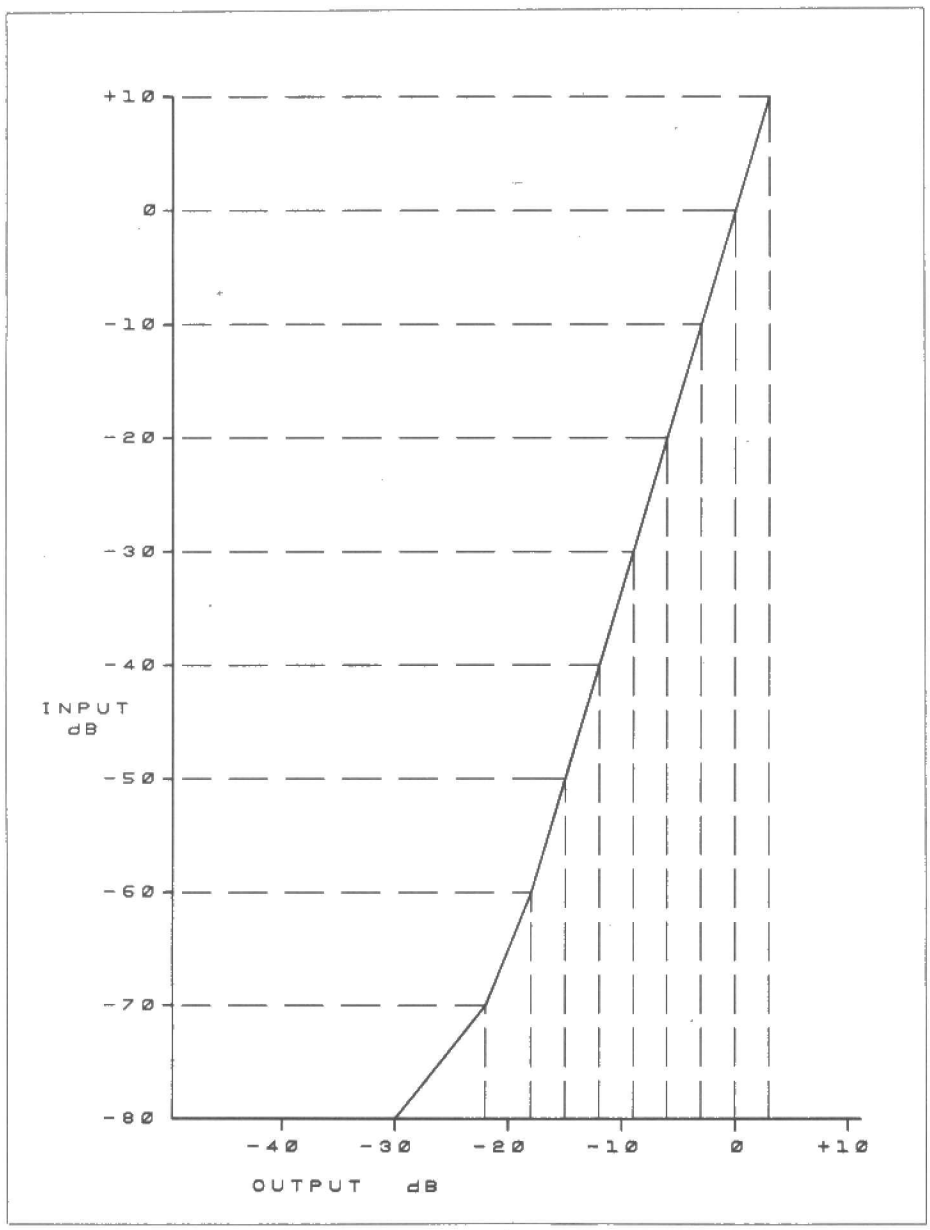

Figure 6. Input/output transfer characteristics.

positioning each item, see Figure 8. The majority of the components are supplied in the main kit (LM60Q), but some RF components have to be selected from the appropriate tuning pack for the desired band, see Table 1.

The sequence in which the components are fitted is not critical. However, the following instructions will be of use in making these tasks as straightforward as possible. It is usually easier to start with the smaller components, such as the resistors. Next mount the ceramic, polyester, polystyrene and electrolytic capacitors. The polarity for the electrolytic capacitors is shown by a plus sign (+) matching that on the PCB legend. However on some capacitors the polarity is designated by a negative symbol (−), in which case the lead nearest this symbol goes away from the positive sign on the legend. All the silicon diodes have a band at one end. Be sure to position them according to the legend, where the appropriate markings are shown. The three varicaps VC1, VC2 and VC3 resemble transistors and you must carefully match the case to the outline shown on the legend. Next install all the transistors, matching each case to its outline. The dual gate MOSFET transistor TR1 is a surface mounting component and is shown in Figure 9. When fitting the IC sockets ensure that you install the appropriate one at each position, matching the notch with the block on the legend. Install IC1 to IC7 making certain that all the pins go into their sockets and the pin one marker is at the notched end. The RF transformers T1, 2, 3 and T4 will only fit one way, but make certain that they are pushed down firmly on to the surface of the PCB. Next install the three preset resistors RV1, 5, 6 and the two trimmer capacitors TC1, 2 and set them all to their half way positions. Install pins at the test points TP1 to TP6 ensuring that you push them fully into the board. When fitting the 'Minicon' connectors ensure that the locking tags are facing inwards, see Photo 1. Finally, mount the two voltage regulators, RG1 and RG2 in their correct positions on the board, ensuring

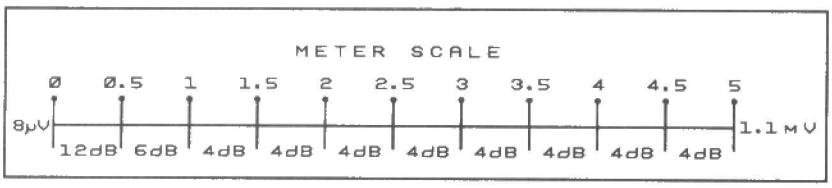

Figure 7. Signal strength meter response.

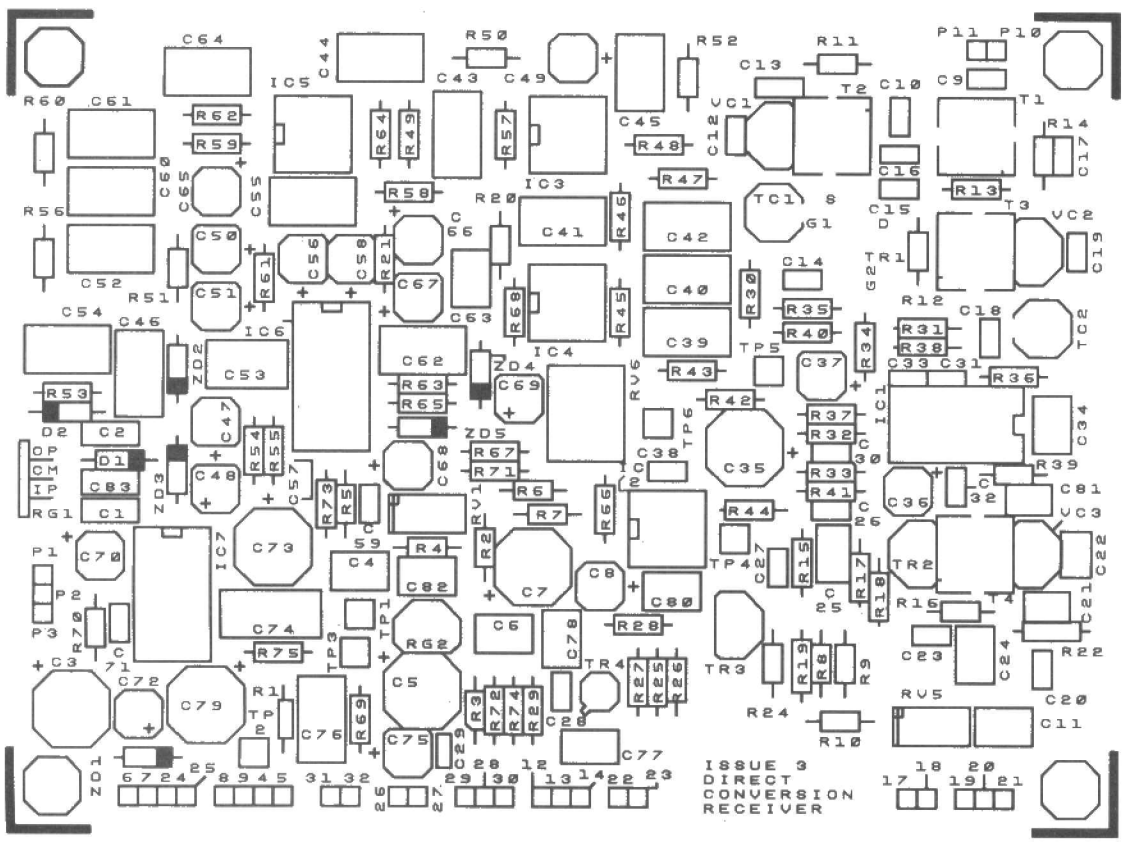

Figure 8. PCB layout.

that you fit RG1 as shown in Figure 15.

This completes the assembly of the PCB and you should now check your work very carefully making sure that all the solder joints are sound. It is also very important that the solder side of the circuit board does not have any trimmed component leads standing proud by more than 3mm, as this may result in a short circuit. Further information on soldering and assembly techniques can be found in the 'Constructors Guide' included in the kit. Photo 2 shows the completed PCB in clear detail.

Final Assembly

The unit is designed to fit in to a metal case type 2408/160 (Maplin code XJ33L) which is also available ready drilled (Maplin code YT05F). An internal chassis is also available for this box (XJ41U) and indeed this has been incorporated in the design, once again a ready drilled version is available (YT04E). However, if you wish to make up your own box, drilling details for the box and the chassis are given in Figure 10. Also shown in Figure 10 are details of the potentiometer mounting bracket should you wish to make your own, this item too is available ready made (JG47B). Custom made stick-on panels can be purchased to enhance the final look. When fitting ensure that the front and back metal panels of the box are clean then remove the protective backing from the self-adhesive decorative trims. Carefully position and firmly push down using a dry, clean cloth until the trims are securely in place. Photo 3 shows the rear trim.

Before fitting the headphone jack JK2 remove its forward facing locating tag. When installing the two jack sockets, position a pot washer between the steel chassis and the back of the front panel, see Figure 11. Next prepare the four rotary potentiometers by cutting the shafts to a length of 12mm. When mounting the AF, RF gain and fine tuning pots use two nuts as shown in Figure 12. Before mounting the main tuning control, rotate its shaft to its fully clockwise position. Then back it off a small amount to set the wiper onto the start of the active part of its carbon track. Set the vernier dial to read 100 and remove the small bolt at the rear of the dial. Do not remove

Amateur Band	Receiver Tuning Range	Tuning Pack	T1,2,3,4	1% 0.6W Metal Film			Ceramic Plate			Polystyrene
				R8	R9	R10	C10	C18	C12,13,17,19	C21,22
160M 1.810 – 2.000MHz	1.800 – 2.010MHz	1	KANK 3333R	22k	NOT FITTED	22k	180pF	100pF	120pF	100pF
80M 3.500 – 3.800MHz	3.490 – 3.810MHz			1k	4k7	82k	47pF	27pF	33pF	22pF
40M 7.000 – 7.100MHz	6.690 – 7.150MHz	2	KANK 3334R	10k	NOT FITTED	180k	180pF	27pF	120pF	100pF
10.100 – 10.150MHz	10.000 – 10.500MHz			1k	2k2	150k	47pF		47pF	33pF
20M 14.000 – 14.350MHz	13.990 – 14.400MHz	3	KANK 3335R	1k	4k7	180k	220pF	27pF	100pF	100pF
18.068 – 18.168MHz	18.000 – 18.500MHz				2k2	220k	180pF	15pF	68pF	68pF
15M 21.000 – 21.450MHz	20.990 – 21.500MHz					330k	100pF		47pF	47pF
24.890 – 24.990MHz	24.540 – 25.000MHz	4	KANK 3335R	1k	2k2	560k	56pF	15pF	22pF	22pF
10M 28.000 – 29.700MHz	A, 27.975 – 28.525MHz B, 28.475 – 29.025MHz C, 28.975 – 29.525MHz D, 29.475 – 30.025MHz						27pF		15pF	22pF

Table 1. Band Chart.

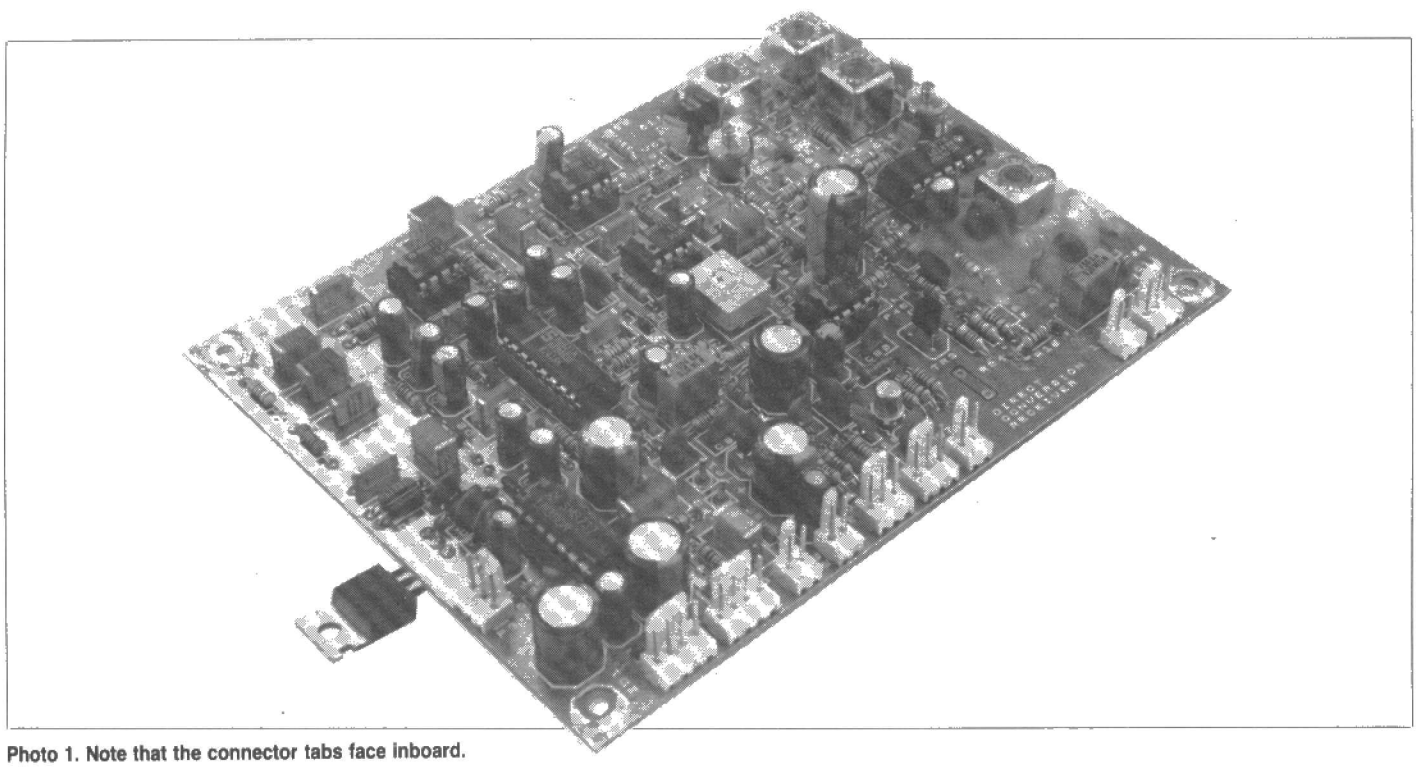

Photo 1. Note that the connector tabs face inboard.

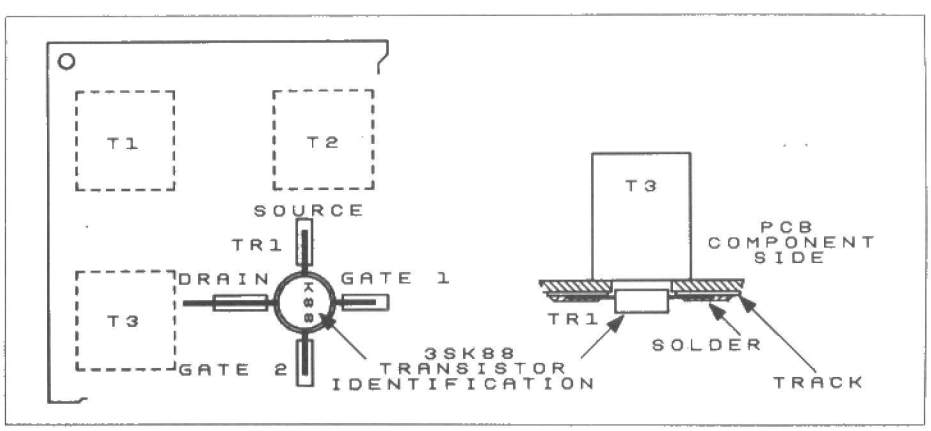

Figure 9. Mounting the dual gate mosfet.

the pot shaft retaining bolt, merely slacken it off enough to allow the shaft to enter the dial. Using M3 hardware, mount the bracket assembly and vernier dial to the front panel, as shown in Figure 13. Next, install the rest of the front panel components and secure the three knobs so that the pointers are at zero when the controls are set fully-anticlockwise.

The EARTH terminal is mounted directly onto the back panel, with the insulating bushes, solder tag and one 4mm washer being discarded, see Figure 14. Fit the two RF connectors; SK3 is the BNC and SK5 is the SO239, ensuring that both are tightly secured. The two 2.5mm power sockets are held in place using M2 bolts and do not require fixing nuts as the mounting holes are threaded.

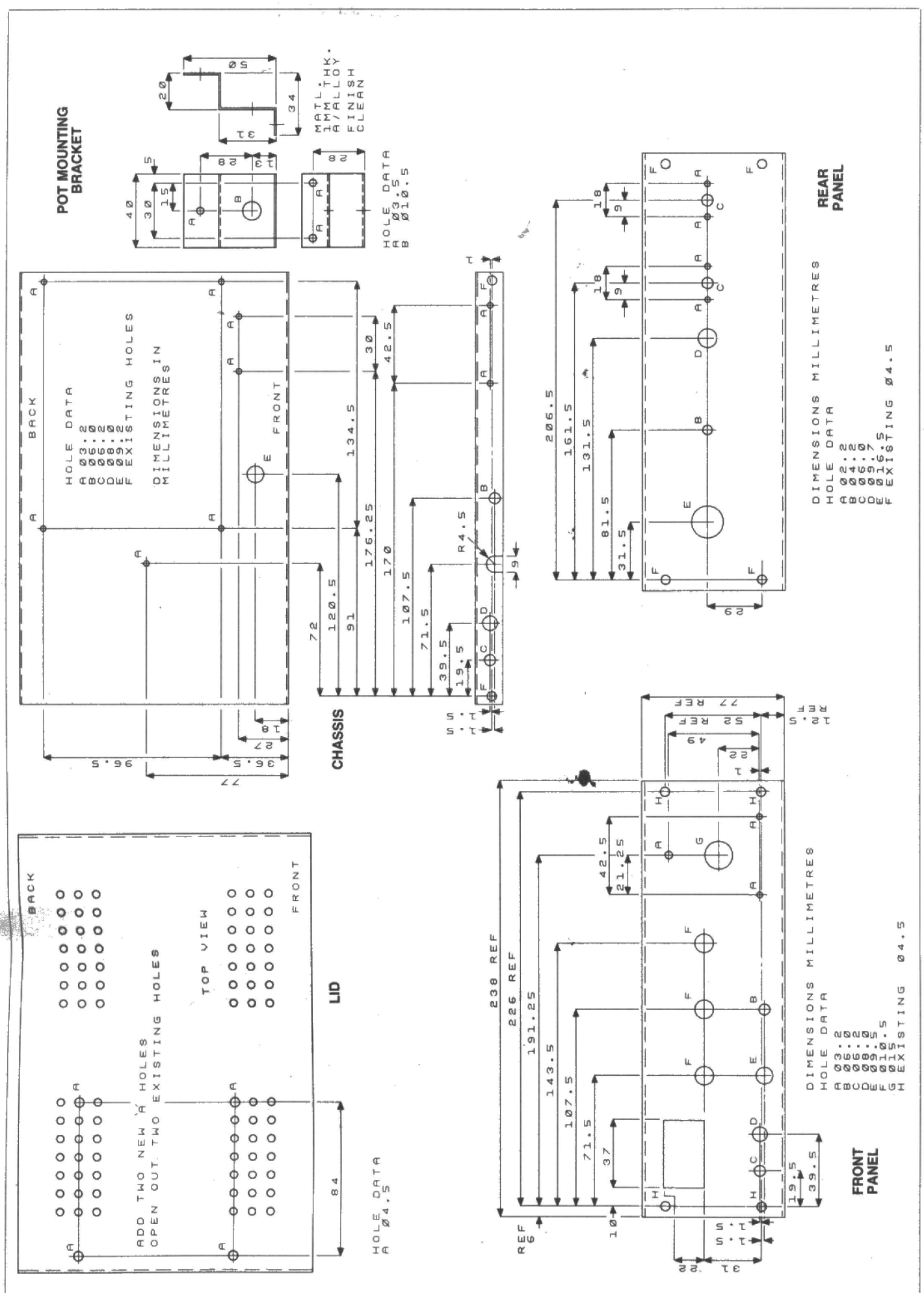
POT MOUNTING BRACKET
MATL. THK. 1MM A/ALLOY FINISH CLEAN
HOLE DATA A Ø3.5 B Ø10.5
CHASSIS
HOLE DATA A Ø3.2 B Ø6.2 C Ø8.2 D Ø8.2 E EXISTING HOLES
DIMENSIONS IN MILLIMETRES
BACK
FRONT
REAR PANEL
DIMENSIONS MILLIMETRES
HOLE DATA A Ø2.2 B Ø4.2 C Ø6.7 D Ø16.5 F EXISTING Ø4.5
LID
ADD TWO NEW 'A' HOLES
OPEN OUT TWO EXISTING HOLES
TOP VIEW
HOLE DATA A Ø4.5
FRONT PANEL
DIMENSIONS MILLIMETRES
HOLE DATA A Ø3.2 B Ø6.2 C Ø8.2 D Ø9.5 E Ø10.5 F Ø15 G Ø15 H EXISTING Ø4.5

Figure 10. Mounting bracket and box drilling details.

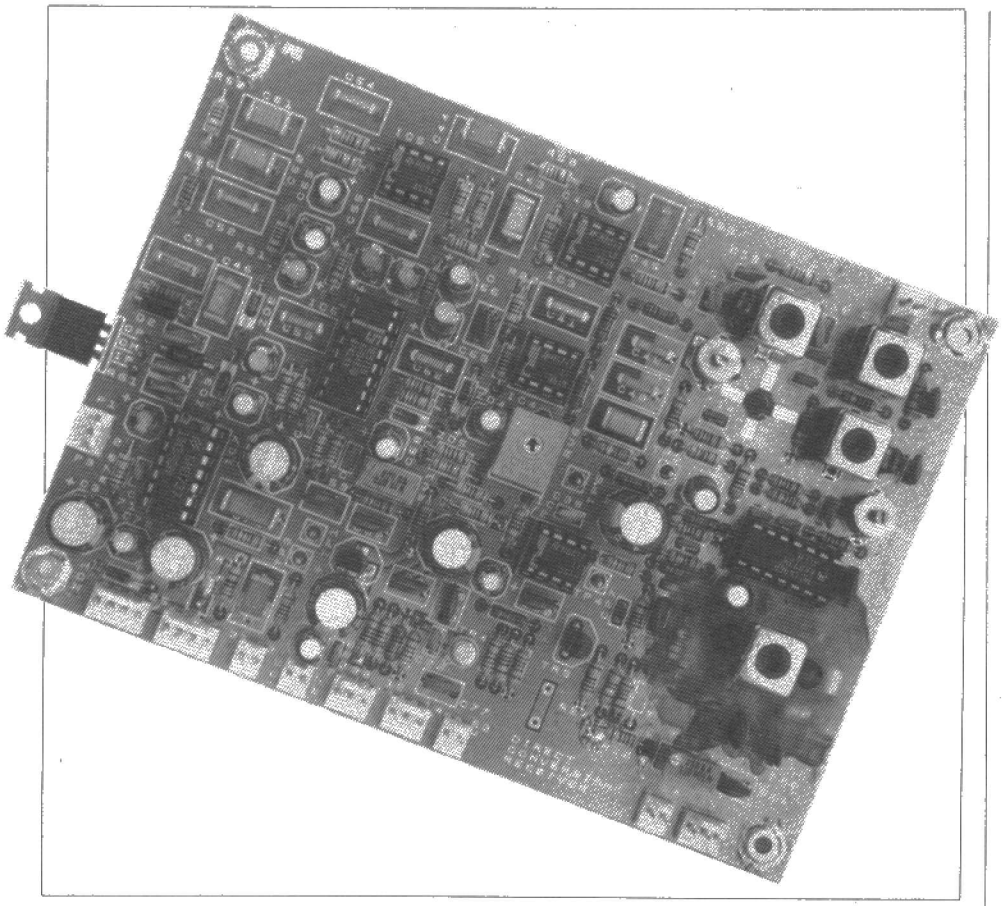

Photo 2. The completed PCB.

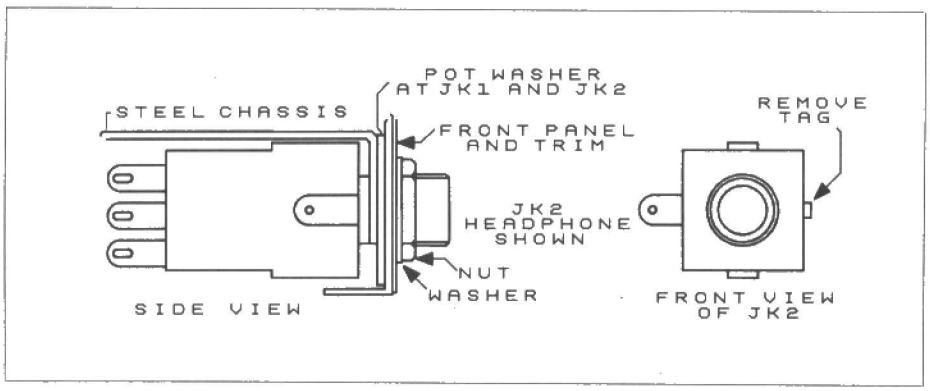

Figure 11. Mounting JK1 and JK2.

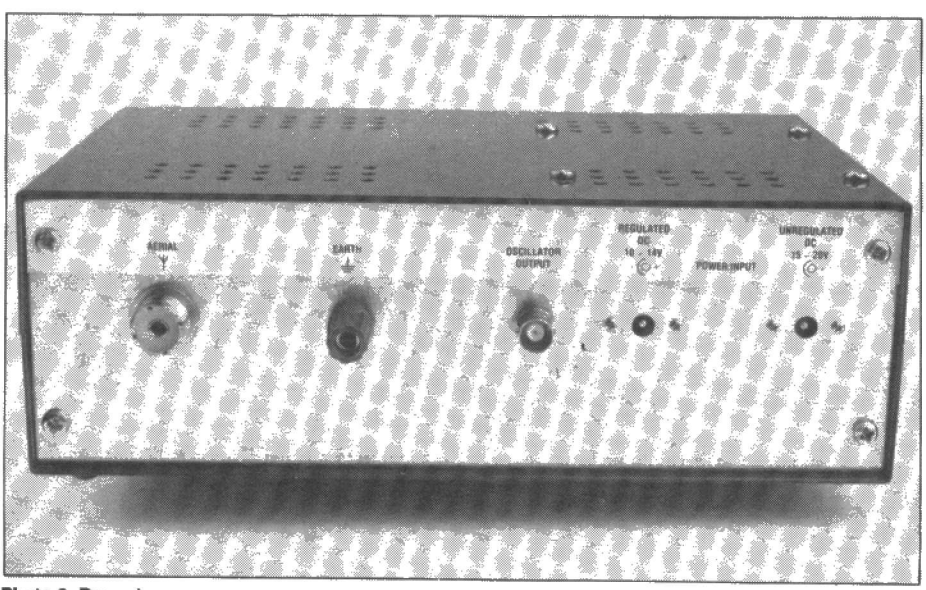

Photo 3. Rear view.

The PCB assembly is secured to the steel chassis using 6BA hardware, see Figure 15. When mounting the voltage regulator RG1 ensure that the mica washer and insulating bush are correctly positioned. It is not essential to use heat transfer compound on this device as it is run well within its safe working limits.

The loudspeaker is mounted to the top lid of the case using M4 hardware, as shown in Figure 16. The nut between the speaker and the case is used as a spacer to assist the sound output. This completes the mechanical assembly of the unit and you should now check your work very carefully before proceeding to the wiring stage.

Wiring

If you purchase a complete kit from Maplin it should contain six, one metre lengths of coloured hook-up wire. No specific colour has been designated for each wire connection, it is entirely up to you. The use of coloured wire is to simplify matters, thus making it easier to trace separate connections to off-board components, just in case there is a fault in any given part of the circuit. A wiring diagram showing all the interconnections is given in Figure 17. The wire connections to the PCB are made using 'Minicon' connectors and the method of installing them is shown in Figure 18. All the wires to the front panel components below the internal chassis pass through a small PVC grommet, see Photo 4. The connections to the headphone jack socket JK2 and the loudspeaker LS1 are made using 'figure 8' zip cable. Do not forget to fit the two link wires or the resistors, R76 and R77 on JK2. When wiring the filament bulb LP1 follow the assembly and mounting instructions in Figure 19 which shows construction details for a cardboard shroud which is used to mount the bulb on the rear of the meter, see Photo 5.

The cable used on SK3 and SK5 has a semi-conducting polythene sheath between the braided screen and the inner insulated conductor. It is most important that the semi-conducting sheath should not be able to come into contact with the centre conductor or anything connected to the centre conductor,

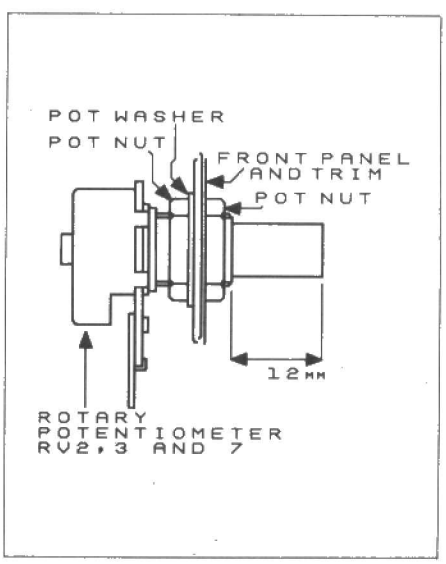

Figure 12. Mounting the AF/RF gain and Fine Tuning pots.

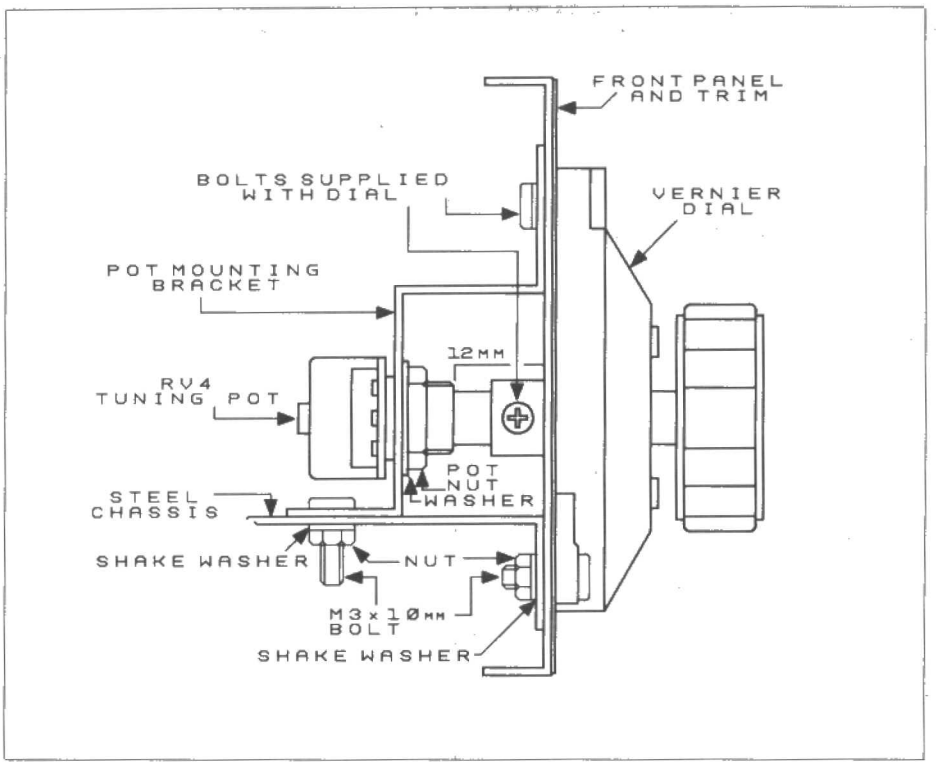

Figure 13. Mounting the Tuning pot.

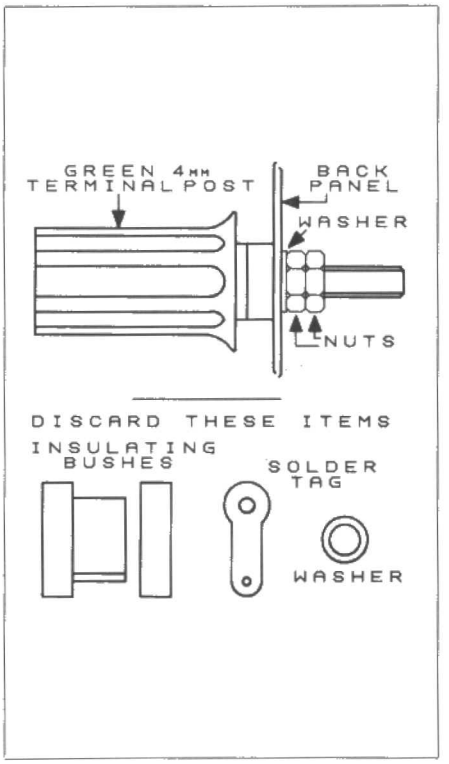

Figure 14. Fixing the earth terminal.

but that it should be stripped back to the braiding. Photo 6 shows the connections to SK3 and SK5.

This completes the wiring of the receiver and you should now check your work very carefully making sure that all the solder joints are sound.

Testing and Alignment

Before you commence testing the unit set the PCB presets and the front panel controls to the following positions:

On the PCB

RV1, RV5 and RV6 should be set to their half way positions. RV1 and RV5 are 22-turn cermet presets with a slipping clutch which should make an audible clicking sound at each end of their travel. To set them half way, simply rotate the adjustment screw until the clicking sound is heard, then reverse the direction for 11 turns.

On the Front Panel

Set the AF and RF gain controls at zero with the fine tuning set half way to its zero mark. Set the vernier to read 100 on its dial and ensure that the power switch is in the OFF position.

The power supply used during the test procedure was an unregulated 1A mains adaptor type (Maplin code YM85G) and it was set to its 12V output. The polarity switch must be set for positive (+) on the tip of the plug, otherwise damage may be caused to the receiver and power supply when connected. If you intend to use other power supplies, such as rechargeable lead-acid or nickel cadmium batteries, it is STRONGLY recommended that an in-line 1 amp fuse be placed in the positive power line. DO NOT connect the power supply until it is called for during the test procedure.

The aerial used was a 30 metre end fed long wire at approximately 10 metres above the ground. However, the aerial you use will be determined by the available space and the amateur band you have chosen. The theoretical and practical design of shortwave aerials is an involved topic which can not be covered by this article. However, a list of some of the many books available on this subject is given at the end. DO NOT connect the aerial to the receiver until it is called for.

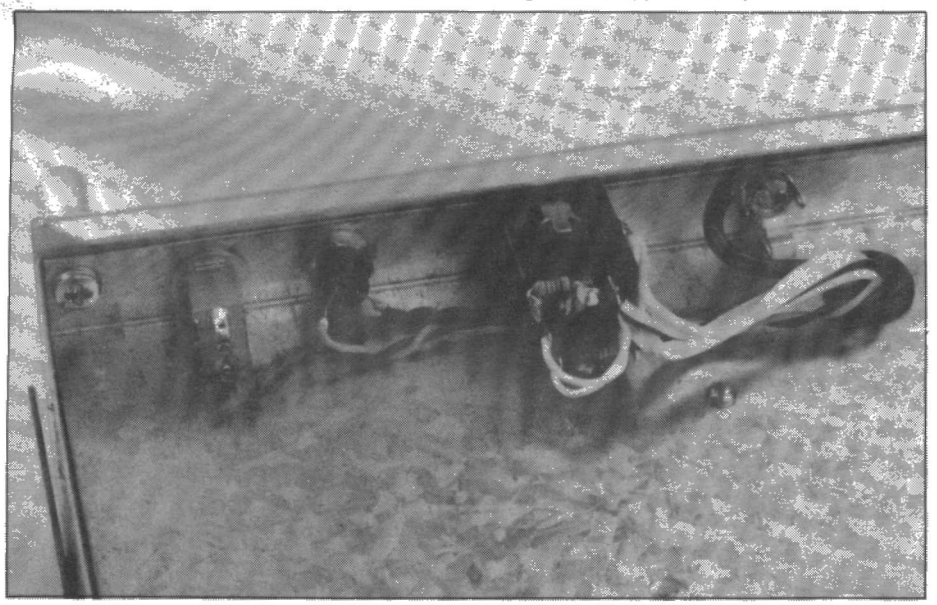

Photo 4. Under the chassis.

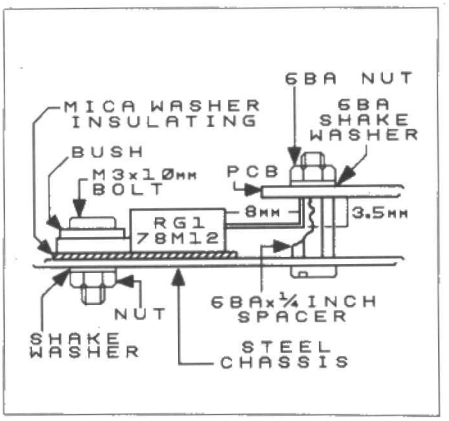

Figure 15. Mounting the PCB and Regulator.

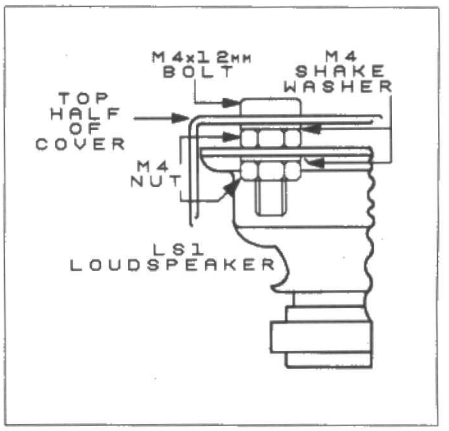

Figure 16. Loudspeaker fixing.

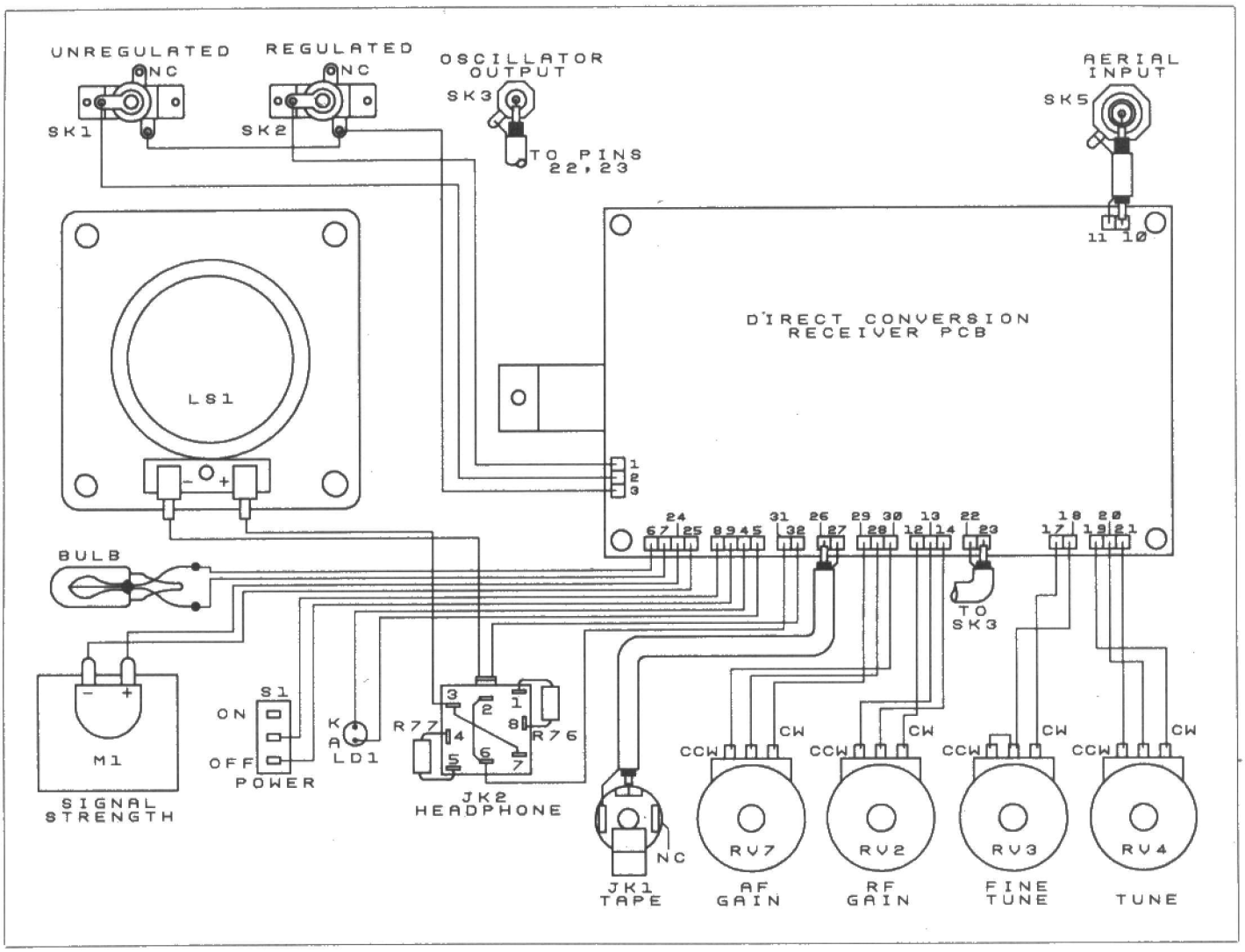

Figure 17. Wiring.

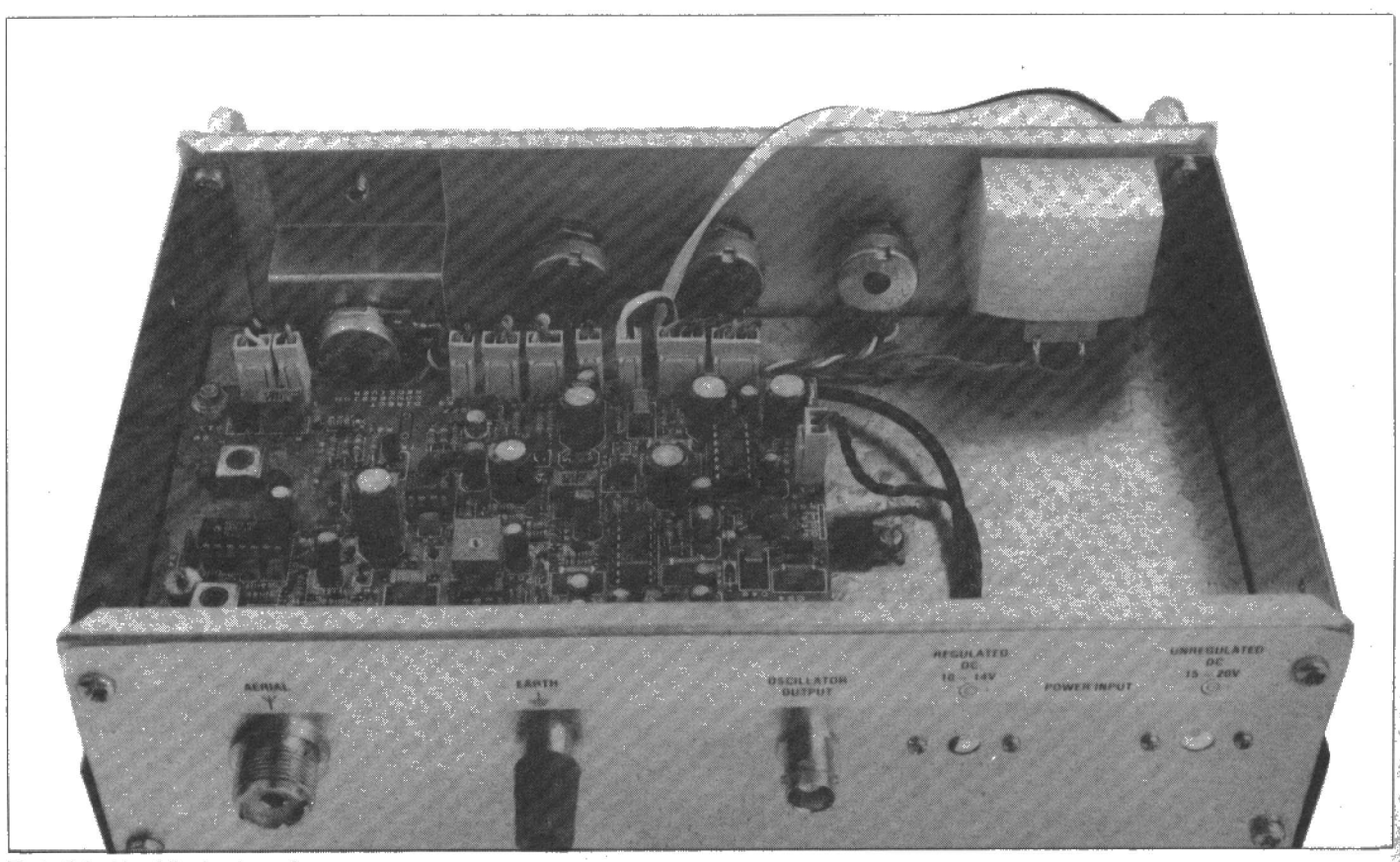

Photo 5. Inside of the front panel.

To measure the DC voltage, current and resistance you will need a digital, or electronic analogue, multimeter with an input impedance of not less than 10MΩ. Other multimeters can be used but some of the readings obtained may vary slightly.

The first test is to measure the resistance at SK1 and SK2. With the meter leads either way round on the centre pin of the unregulated power input socket SK1, a reading of greater than 100kΩ should be present with respect to the chassis. This reading should drop by a small amount when the power on/off switch S1 is activated. Repeat this test on the regulated input SK2, where you should observe the following readings. With the power switch in its OFF position and the test meter leads either way round on the centre pin, an infinite resistance 'open circuit' should be present with respect to the chassis. However, when switched ON this reading should fall to approximately 17Ω. This is mainly due to the low resistance of the filament bulb LP1 and if the Minicon connector in front of ZD1 is temporarily disconnected this reading should rise to greater than 1kΩ if the positive probe is on the centre pin of SK2.

Select a suitable range on your meter that will accommodate a 500mA DC current reading and place it in the positive power line from SK1. Connect the mains adaptor and switch on. The signal strength meter and power indicator LD1 should light up, with a current reading of approximately 147mA being observed. Switch off the receiver, then remove the test meter and reconnect the positive line to SK1.

Now set your multimeter to read DC volts. All voltages are positive with respect to ground, so connect your negative lead to a convenient ground point on the chassis. When the receiver is switched back on, voltages present on the PCB test points should approximately match the following:

TP1 = 8.67V
TP2 = 12.03V
TP3 = 10.19V
TP4 = 5.09V
TP5 = 11.36V
TP6 = 5.08V

The voltage at TP1 must now be set to exactly 8.000V by adjusting RV1. Whilst observing the signal strength meter, set RV6 so that the needle is just on the beginning of the zero marker. This completes the DC testing and alignment of the receiver, which must now be left running for a minimum of one hour before commencing the RF adjustments.

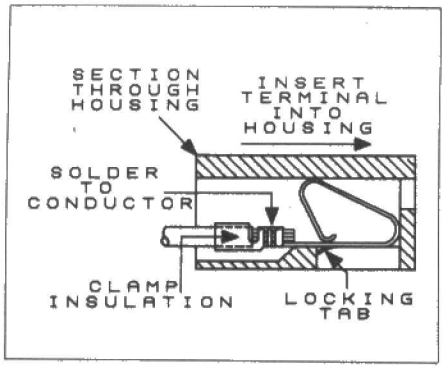

Figure 18. Terminating the wires.

There are three basic methods of RF alignment:

1. The high accuracy approach using an RF signal generator and frequency counter.
2. Medium accuracy using another receiver and aerial on the same band.
3. The 'poke and hope' method using only an aerial.

Method 1

When using a frequency counter to set the tuning range, connect its input to the oscillator output on the back of the receiver and use as short a lead as possible to prevent stray RF pick up. Ensure that the main tuning vernier dial is set to 100 and the fine tune is at the half way zero point. Using a non-metallic trimming tool, carefully adjust the iron dust core of T4 until the top end (high frequency) is set. Then position the main dial to read 0 and adjust RV5 for the low end of the band. Repeat this procedure until both limits have been set for the required band, as shown in Table 1.

When using an RF signal generator to align the front end, its output is connected to SK5 the aerial input of the receiver. The level of RF injection should be set to produce a half scale deflection on the signal strength meter when the carrier is tuned in. If the AF gain is turned up, an audio tone should be heard whose frequency will change as the receiver is tuned across the incoming signal. At the low end of the band the dust cores of T2 and T3 should be adjusted to give a peak in signal strength. The top end of the tuning range is then peaked using the trimmer capacitors TC1 and TC2. Next set the tuning dial to 50 and adjust the input filter coil T1 for maximum signal reading. Repeat this procedure until no

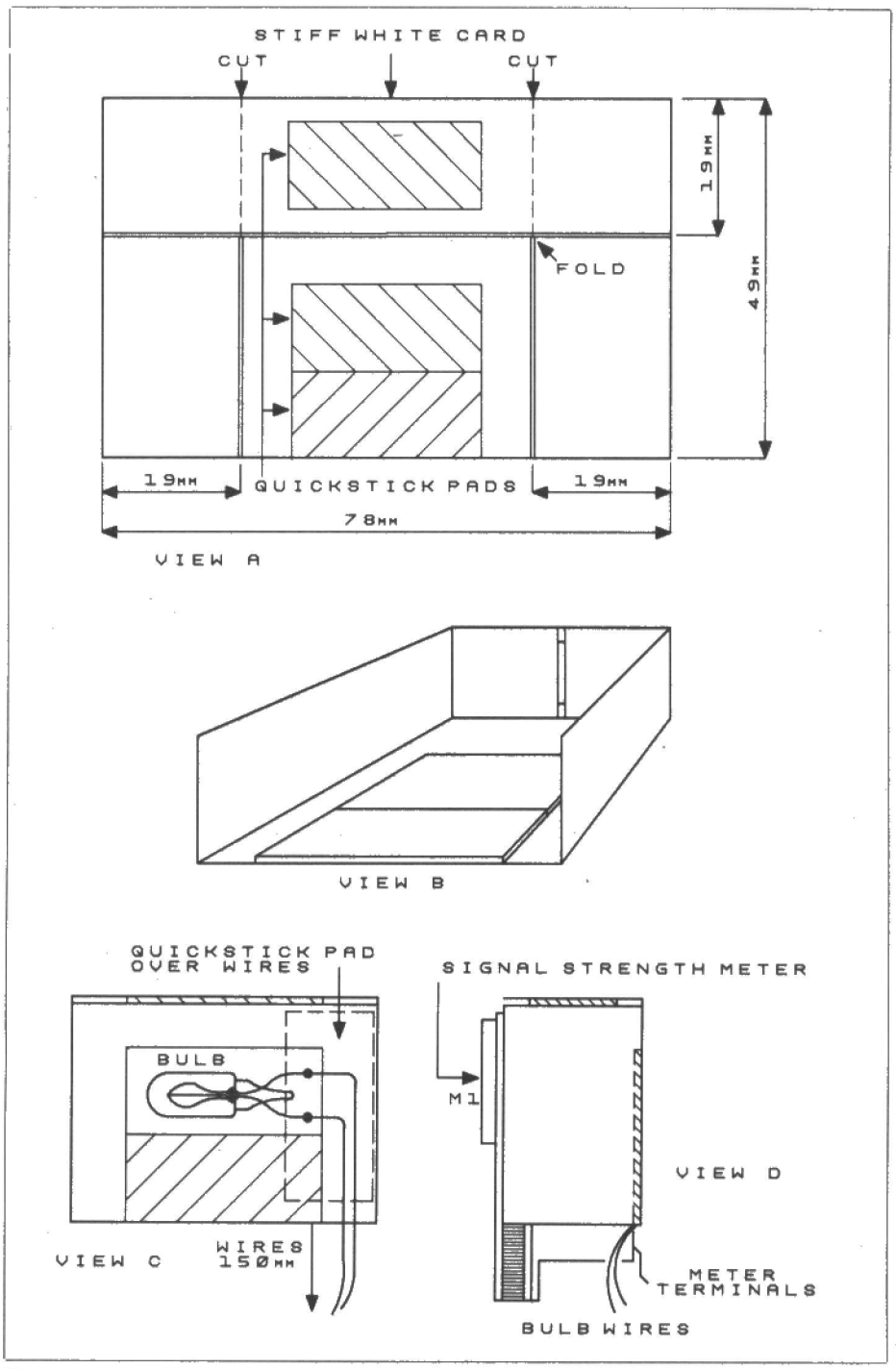

Figure 19. Meter illumination.

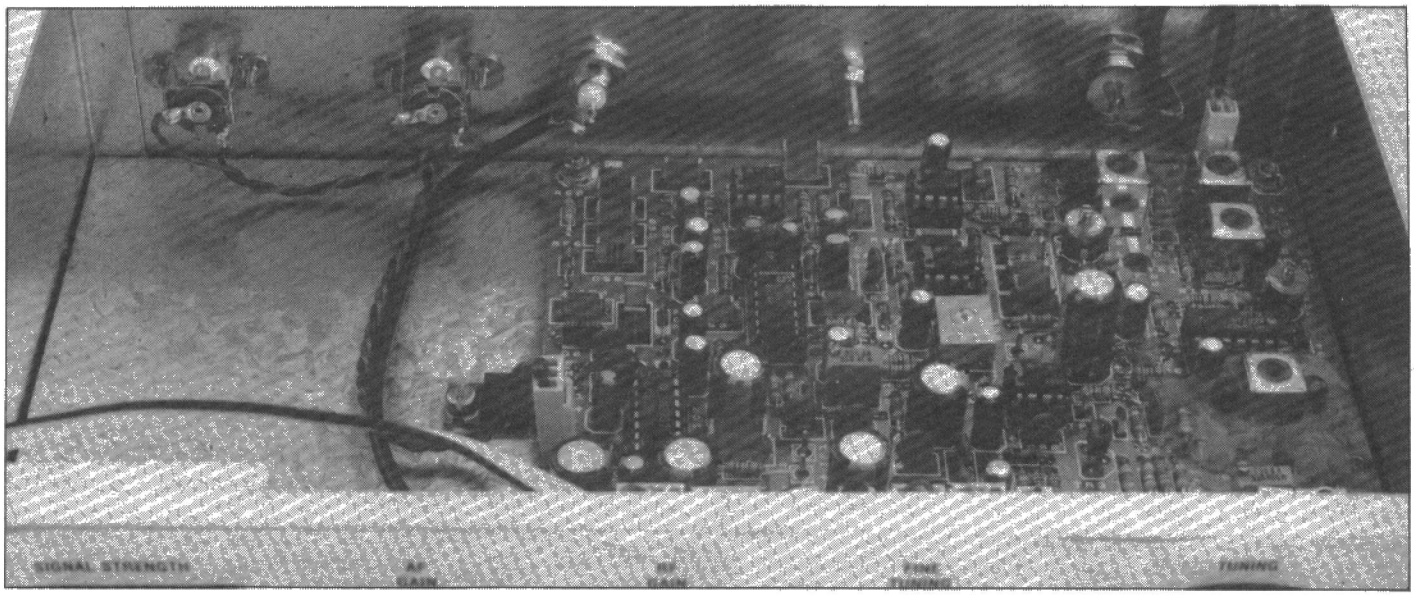

Photo 6. Inside of the rear panel.

further improvement in signal strength can be detected. Finally, check that the RF gain and fine tuning controls are functioning correctly.

Method 2

When using another receiver to set the tuning range of the oscillator no direct connection is made between them. To pick up the RF output, a short piece of wire approximately 200mm (8 inches) long is inserted into the centre terminal of the BNC socket. With a suitable aerial on the other receiver, tune it to the end of the band and listen for the carrier coming from your direct conversion receiver. To adjust this, use T4 for the high and RV5 for the low frequency limit, repeat this procedure until both ends of the band are set.

To align the front end using this method, tune the other receiver into a readable signal at the low end of the band. Swap the aerial over and try to tune for the same station on your direct conversion receiver, adjusting T2 and T3 for a peak in signal strength. Next, put the aerial back on the other receiver and tune for a signal at the high frequency end. Swap back the aerial, tune in the signal adjusting TC1 and TC2 for maximum reading. In a similar manner, set the vernier dial to read approximately 50 and using T1 tune for a peak in signal strength. Repeat these alignment procedures until no further improvement in reception can be obtained.

Method 3

If no test gear or receiver is available, you must set the frequency coverage by trial and error using off air signals. In general, the lower part of the amateur band is used for morse code (CW) transmissions, whilst the upper part is used for speech (SSB). Tuning in an SSB signal properly is quite tricky, but with a little practice it is something that is easily mastered. Very careful tuning is required in order to bring the audio to the correct pitch and the fine tuning control RV3 will make this task much easier. Set the main tuning dial to read 0 and adjust T4 until you start to receive CW transmissions, peaking the signal using T2 and T3. As you tune progressively higher in frequency you should find the SSB transmissions and at the extreme end of the dial TC1 and TC2 are used to peak the signal. If the SSB stations appear to extend past this point, retune T4 until no more amateur transmissions can be received, then reset the low end of the band using RV5. Next, tune for a station near to the centre of the dial (50) and using T1 adjust for maximum reading on the signal strength meter. Repeat these alignment procedures until no further improvement in reception can be obtained.

Once you have successfully completed the RF alignment, the oscillator stage should be covered in candle wax, see Photo 7. Using a lighted household candle drip the molten wax over the oscillator components being very careful not to get any inside the top of T4. Build up the wax a layer at a time and allow it to cool between applications until the components are completely encapsulated. The wax will cause a shift in oscillator tuning and when it has cooled down to room temperature, which may take some time, you must retune T4 using one of the alignment methods previously described.

Using the Receiver

A good aerial and earth system are essential if the receiver is to be expected to perform at its best, and you should allow it to 'warm up' for at least 30 minutes to obtain frequency stability. The skill of the operator will also play a large part in weeding out the more exotic long distance (DX) stations from the interference (QRM/QRN) on today's crowded bands.

There are two main types of radio signal used by amateurs, morse code (CW) and speech (SSB). When tuning in a CW transmission, the pitch of the dots and dashes start high, then as the receiver is tuned closer to the same RF frequency, the audio pitch will fall. At the dead centre there will be no audio tone as the signal is at the zero beat point. But, if the receiver tuning is continued the pitch of the tone will increase until it goes outside the frequency range of the audio bandpass filter. The final pitch setting will be a matter of personal taste, however 500Hz to

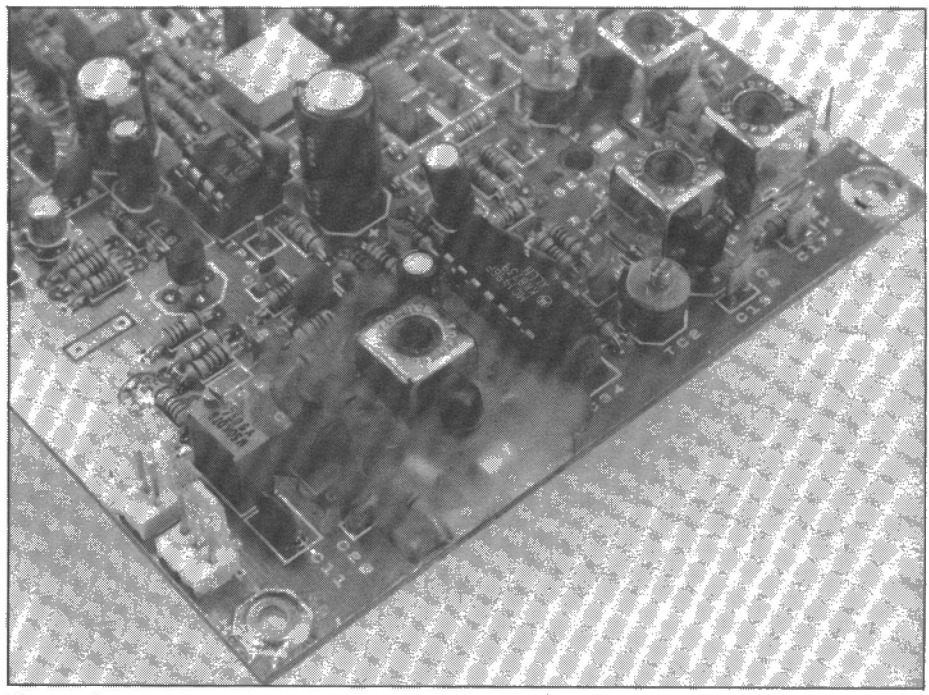

Photo 7. Oscillator section encapsulated in wax.

1kHz is a commonly used range.

The use of single sideband (SSB) for the transmission of speech is preferred by most radio amateurs because of its higher efficiency over ordinary amplitude modulation (AM). In an AM radio signal the level of the RF carrier is modulated by the audio produced by the user. With that type of receiver a small tuning error will probably give no more than a slight loss of audio quality and could even go completely unnoticed. This is not the case when receiving SSB, where the tuning has to be spot on if the station is to be fully resolved. A product of an AM signal is the generation of two sideband signals, lower sideband (LSB) and upper sideband (USB). The sidebands contain the same audio information as the modulated carrier but in a different form. This means that to transmit the information only one of the sidebands needs to be sent and the carrier with the other sideband can be suppressed. However, to resolve the signal the carrier must be re-injected by the product detector in the receiver, which is why the tuning is so critical. The direct conversion receiver principle will permit double sideband signals (DSB) to be received. However, these are not commonly found on the amateur bands where LSB is used up to 40 metres and USB is used on all frequencies above 10MHz. When an SSB station is tuned in on a DSB receiver you will get two places where the signal will be received, but only one sideband will be resolvable. This is the one that matches the original transmitted signal and the overall tone of the audio is set by fine tuning the receiver.

A useful addition to the receiver is a digital frequency counter used to provide an accurate tuning display. The counter should be capable of resolving down to the nearest 100Hz when set to its 0.1 of a second gate time. To prevent the pick up of stray RF signals it should be connected to the oscillator output of the receiver using the shortest possible length of cable. If a frequency counter is not permanently available then a tuning chart can be generated using graph paper, see Figure 20. This will help to provide a reasonable tuning accuracy from the vernier dial when the counter is not available.

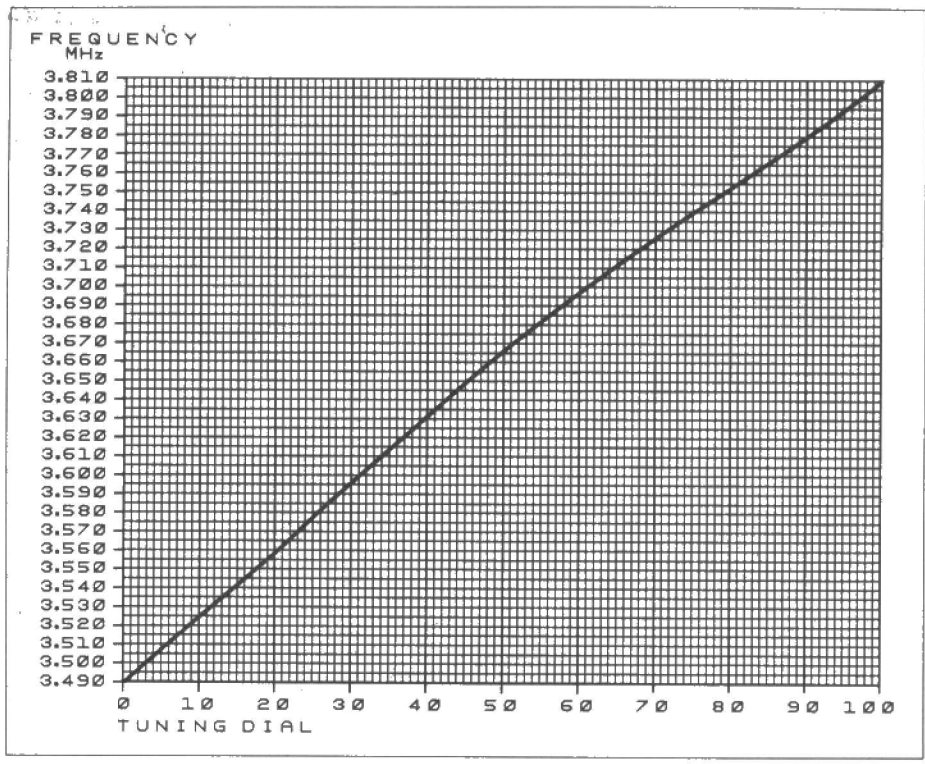

Figure 20. Receiver tuning range (80 metre version).

If the station being received is excessively strong it may start to distort the audio output. This can usually be cured by reducing the RF gain until the reading on the signal strength meter is just below the 5 on the scale.

That wraps it up for this project, I hope you spend many happy hours building your receiver (finished unit shown in Photo 8) and that you find it a pleasure to use once you're on the air.

Useful Reading

HF Antennas by L.A.Moxon, G6XN. Published by the Radio Society of Great Britain.

Wires and Waves. Published by Practical Wireless.

Radio Wave Propagation (HF Bands) by F.C. Judd.

25 Simple Indoor and Window Aerials by E.M. Noll.

An Introduction to Radio DXing by R.A. Penfold.

Beginner's Guide to Amateur Radio by F.G. Rayer.

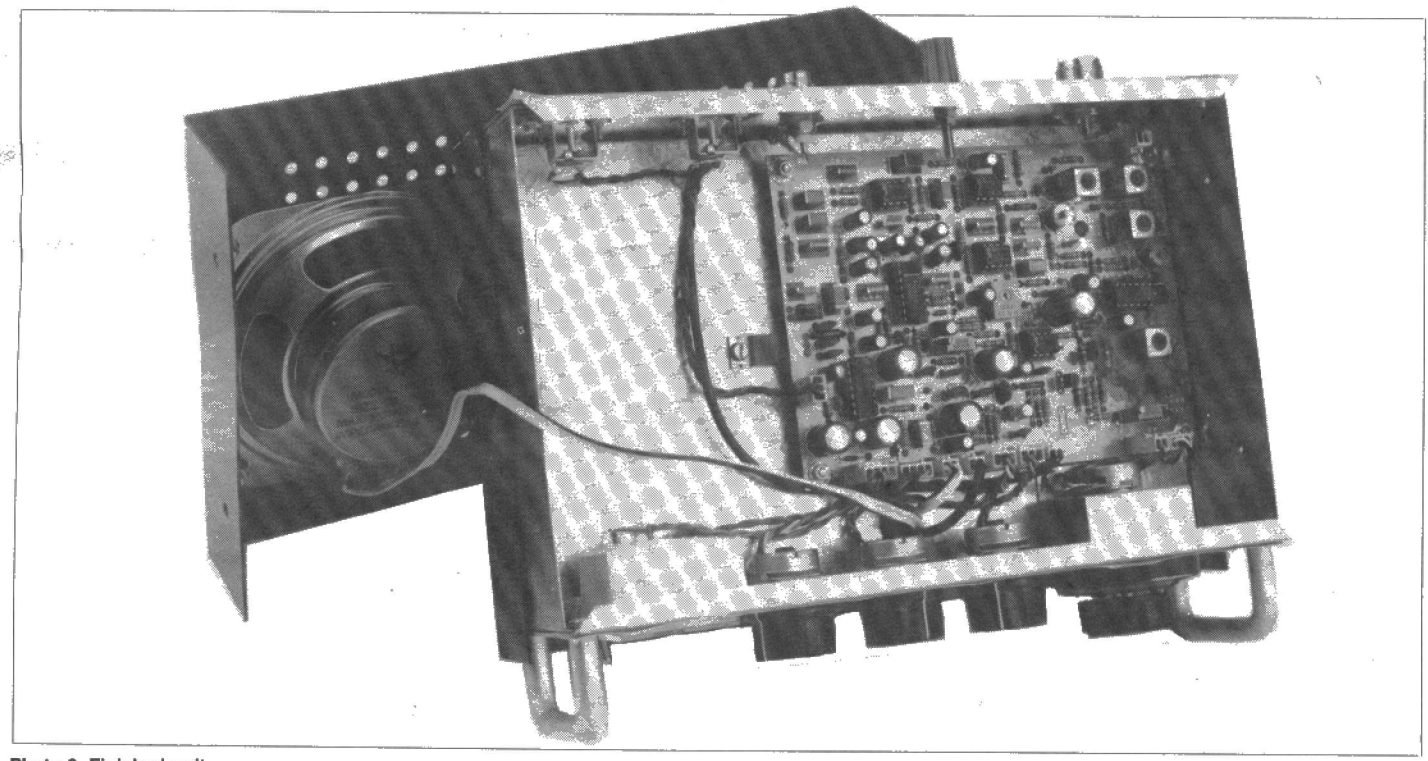

Photo 8. Finished unit.

DIRECT CONVERSION RECEIVER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,33	560Ω	2	(M560R)
R2,76,77	270Ω	3	(M270R)
R3,12,13,30, 36	100Ω	5	(M100R)
R4,15,25,26,31,35, 38,51	1k	8	(M1K)
R5,32	1k2	2	(M1K2)
R6,7,60,66, 68,69	4k7	6	(M4K7)
R17,39,40	2k7	3	(M2K7)
R11,14,18,22, 44,52,64	100k	7	(M100K)
R16	150Ω	1	(M150R)
R19,43	39k	2	(M39K)
R20,21	10M	2	(M10M)
R24,48,58	1M	3	(M1M)
R27	47Ω	1	(M47R)
R28,37,41,42,45, 46,47,53,56,74	10k	10	(M10K)
R29	470Ω	1	(M470R)
R34	820Ω	1	(M820R)
R49,59	6k8	2	(M6K8)
R50,62	3k3	2	(M3K3)
R54,55,57,61, 63,65	27k	6	(M27K)
R67	220Ω	1	(M220R)
R70	150k	1	(M150K)
R71	18k	1	(M18K)
R72	22k	1	(M22K)
R73	1Ω	1	(M1R)
R75	2Ω7	1	(M2R7)
RV1	1k Cermet 22T	1	(UH23A)
RV2,4	47k Pot Lin	2	(FW04E)
RV3	1k Pot Lin	1	(FW00A)
RV5	50k Cermet 22T	1	(UH26D)
RV6	1k Hor Preset	1	(UH00A)
RV7	10k Pot Log	1	(FW22Y)
CAPACITORS			
C1,2,4,6,11,24,25, 34,63,77,78,80, 82,83	100nF Minidisc	14	(YR75S)
C3,5,7,73,79	220μF 16V PC Electrolytic	5	(FF13P)
C8,36,37,48,49,50, 65,66,69,72	1μF 100V PC Electrolytic	10	(FF01B)
C9	22pF Ceramic	1	(WX48C)
C71	47pF Ceramic	1	(WX52G)
C14,15,16,20,23, 29,30,31,32,33, 57,59	10nF Ceramic	12	(WX77J)
C26	1nF Ceramic	1	(WX68Y)
C27	5p6F Ceramic	1	(WX41U)
C28	100pF Ceramic	1	(WX56L)
C35	470μF 16V PC Electrolytic	1	(FF15R)
C38	2n7F Ceramic	1	(WX73Q)
C39,43,44,46,60, 61,76	100nF Polylayer	7	(WW41U)
C40,54	6n8F Polylayer	2	(WW27E)
C41,45,52,64	22nF Polylayer	4	(WW33L)
C42,55	1n5F Polylayer	2	(WW23A)
C47,51,66,67, 68,75	10μF 50V PC Electrolytic	6	(FF04E)
C53,62	10nF Polylayer	2	(WW29G)
C58	2μ2F 100V PC Electrolytic	1	(FF02C)
C70	4μ7F 63V PC Electrolytic	1	(FF03D)
C74	150nF Polylayer	1	(WW43W)
C81	100pF Polystyrene	1	(BX28F)
TC1,2	22pF Trimmer	2	(WL70M)
SEMICONDUCTORS			
IC1	MC1496	1	(QH47B)
IC2,3,5	LF353	3	(WQ31J)
IC4	LF351	1	(WQ30H)
IC6	NE571	1	(YY87U)
IC7	LM380	1	(QH40T)
D1,2	1N4001	2	(QL73Q)
TR1	3SK88	1	(UH63T)
TR2,3	BF244	2	(QF16S)
TR4	BC179	1	(QB54J)
VC1,2,3	BB212 Varicap	3	(YH83E)
RG1	78M12UC	1	(QL29G)
RG2	LM317LZ	1	(QL87U)
ZD1	BZY88C4V3	1	(QH05F)
ZD2,3,4,5	BZY88C2V7	4	(QH00A)
LD1	LED Red	1	(QY48C)
MISCELLANEOUS			
S1	Sub-min Toggle A	1	(FH00A)
M1	Signal Strength Meter	1	(LB80B)
LS1	4in. Spkr 8Ω	1	(YJ16S)
	Pin 2148	1 Pkt	(FL24B)
	P.C. Board	1	(GD78K)
LP1	Wire Bulb 12V	1	(WQ13P)
SK1,2	Power Socket 2.5mm	2	(HH86T)
SK3	BNC Socket 50Ω	1	(HH18U)
SK4	Terminal Post Green 4mm	1	(HF05F)
SK5	Socket SO239	1	(BW84F)
JK1	Jack Socket 3.5mm	1	(HF82D)
JK2	Switched Jack Socket ¼in.	1	(BW80B)
	Vernier Dial Ratio 7·5:1	1	(RX40T)
	DIL Socket 8-pin	4	(BL17T)
	DIL Socket 14-pin	2	(BL18U)
	DIL Socket 16-pin	1	(BL19V)
	Minicon Latch Plug 2W	5	(RK65V)
	Minicon Latch Plug 3W	4	(BX96E)
	Minicon Latch Plug 4W	2	(YW11M)
	Minicon Latch Housing 2W	5	(HB59P)
	Minicon Latch Housing 3W	4	(BX97F)
	Minicon Latch Housing 4W	2	(HB58N)
	Minicon Terminal	3 Pkts	(YW25C)
	Kit P Plas	1	(WR23A)
	Knob K7B	3	(YX02C)
	Hook-up Wire Black	1 Pkt	(BL00A)
	Hook-up Wire Blue	1 Pkt	(BL01B)
	Hook-up Wire Green	1 Pkt	(BL04E)
	Hook-up Wire Red	1 Pkt	(BL07H)
	Hook-up Wire White	1 Pkt	(BL09K)
	Hook-up Wire Yellow	1 Pkt	(BL10L)
	Zip Wire	1 Mtr	(XR39N)
	Low Noise Screened Cable	1 Mtr	(XR18U)
	Quickstick Pads	1 Stp	(HB22Y)
	Constructor's Guide	1	(XH79L)
OPTIONAL			
	Power Supply Unregulated	1	(YM85G)
	Power Plug 2.5mm	2	(HH63T)
	Fuse Holder In-line	1	(RX51F)
	Fuse 1.25in. 1A	1	(WR11M)
	Trim Tool Set	1	(BK34M)
	Preset Trim Tool	1	(BK49D)
	Pot Nut M10	1 Pkt	(FP06G)
	Pot Washer M10	1 Pkt	(FP07H)
	Grommet Small	1	(FW89P)
	Box Pre-drilled	1	(YT05F)
	Chassis Pre-drilled	1	(YT04E)
	Pot Mounting Bracket	1	(JG47B)
	Front Panel Stick-on	1	(JG48C)
	Back Panel Stick-on	1	(JG49D)
	Spacer Tapped 6BA x ¼in.	1 Pkt	(FD10L)
	Bolt 6BA x ½in.	1 Pkt	(BF06G)
	Nut 6BA	1 Pkt	(BF18U)
	Shake Washer 6BA	1 Pkt	(BF26D)
	Isobolt M2 x 6mm	1 Pkt	(JD11M)
	Isobolt M3 x 10mm	1 Pkt	(HY30H)
	Isonut M3	1 Pkt	(BF58N)
	Isoshake M3	1 Pkt	(BF44X)
	Isobolt M4 x 12mm	1 Pkt	(BF49D)
	Isonut M4	1 Pkt	(BF57M)
	Isoshake M4	1 Pkt	(BF43W)

The parts listed above, excluding Optional, are available as a kit, but is not shown in our 1988 catalogue:
Order As LM60Q (Direct Conversion Receiver Kit) Price £69.95
The following items are also available separately:
Box Pre-drilled **Order As YT05F Price £17.95**
Chassis Pre-drilled **Order As YT04E Price £3.95**
Pot Mounting Bracket **Order As JG47B Price 78p**
Front Panel **Order As JG48C Price £1.98**
Rear Panel **Order As JG49D Price 60p**
Direct Convrsn Rec PCB **Order As GD78K Price £11.95**

PARTS LIST TUNING KIT 1

RESISTORS: All 0.6W 1% Metal Film			
	1k	1	(M1K)
	4k7	1	(M4K7)
	22k	2	(M22K)
	82k	1	(M82K)
CAPACITORS			
	27pF Ceramic	1	(WX49D)
	33pF Ceramic	4	(WX50E)
	47pF Ceramic	1	(WX52G)
	100pF Ceramic	1	(WX56L)
	120pF Ceramic	4	(WX57M)
	180pF Ceramic	1	(WX59P)
	22pF Polystyrene	2	(BX24B)
	100pF Polystyrene	2	(BX28F)
MISCELLANEOUS			
	RF Transformer KANK 3333R	4	(FD02C)

PARTS LIST TUNING KIT 2

RESISTORS: All 0.6W 1% Metal Film			
	1k	1	(M1K)
	2k2	1	(M2K2)
	10k	1	(M10K)
	150k	1	(M150K)
	180k	1	(M180K)
CAPACITORS			
	27pF Ceramic	1	(WX49D)
	47pF Ceramic	5	(WX52G)
	120pF Ceramic	4	(WX57M)
	180pF Ceramic	1	(WX59P)
	33pF Polystyrene	2	(BX25C)
	100pF Polystyrene	2	(BX28F)
MISCELLANEOUS			
	RF Transformer KANK 3334R	4	(FD03D)

PARTS LIST TUNING KIT 3

RESISTORS: All 0.6W 1% Metal Film			
	1k	1	(M1K)
	2k2	1	(M2K2)
	4k7	1	(M4K7)
	180k	1	(M180K)
	220k	1	(M220K)
	330k	1	(M330K)
CAPACITORS			
	15pF Ceramic	1	(WX46A)
	27pF Ceramic	1	(WX49D)
	47pF Ceramic	4	(WX52G)
	68pF Ceramic	4	(WX54J)
	100pF Ceramic	4	(WX56L)
	180pF Ceramic	1	(WX59P)
	220pF Ceramic	1	(WX60Q)
	47pF Polystyrene	2	(BX26D)
	68pF Polystyrene	2	(BX27E)
	100pF Polystyrene	2	(BX28F)
MISCELLANEOUS			
	RF Transformer KANK 3335R	4	(FD04E)

PARTS LIST TUNING KIT 4

RESISTORS: All 0.6W 1% Metal Film			
	1k	1	(M1K)
	2k2	1	(M2K2)
	560k	1	(M560K)
CAPACITORS			
	15pF Ceramic	5	(WX46A)
	22pF Ceramic	4	(WX48C)
	27pF Ceramic	1	(WX49D)
	56pF Ceramic	1	(WX53H)
	22pF Polystyrene	2	(BX24B)
MISCELLANEOUS			
	RF Transformer KANK 3335R	4	(FD04E)

All 4 Tuning Kits are available, but are not shown in our 1988 catalogue:

Order As LM61R (Tuning Kit 1) Price £2.95
Order As LM62S (Tuning Kit 2) Price £2.95
Order As LM63T (Tuning Kit 3) Price £2.95
Order As LM64U (Tuning Kit 4) Price £2.95

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The Editor, Maplin Electronics, P.O. Box 3, Rayleigh, Essex SS6 8LR

LOUDSPEAKERS
THE FIRST 111 YEARS

Part 1: The early years

by J.M. Woodgate B.Sc (Eng), A.M.I.E.E., M.A.E.S., M.Inst. S.C.E.

In 1877, patents for moving-coil loudspeakers were issued to E.W. Siemens in Germany, and, almost simultaneously, to Charles Cuttriss and Jerome Redding in the U.S.A. However, this type of loudspeaker had to wait for more than forty-eight years before becoming widely used, and its invention was both preceded and followed by an enormous number of other inventions, some of which actually worked. Nevertheless, the history of the loudspeaker can be considered to begin in the year of this patent. Even today, loudspeakers are one of the most fascinating subjects for the private inventor, and the flow of ingenious (but usually unproven and/or ineffective) devices from dedicated enthusiasts continues unchecked.

Pre-history

Before 1877, most of the eminent physicists of the nineteenth century had investigated, whether purposely or as a side-issue, the production of sound by electrical means. Although it is possible to do this in several ways, almost all the work was based on the electromagnet, invented by Sturgeon, and the dynamo, invented by Michael Faraday. If d.c. is applied to an electromagnet fitted with a springy armature, a most satisfactory clicking sound can be produced as the circuit is made and broken. This, in fact, is the 'Morse sounder', which featured prominently in early Western films. Faraday's early dynamo produced alternating current (before he invented the commutator, which is actually a synchronous mechanical rectifier), and applying alternating current from a hand-turned dynamo to the aforesaid electromagnet CAN produce an unattractive squalling noise. The reason for the emphasis on 'can' is that the actual sound output from different combinations of dynamo and electromagnet varies vastly, and the reasons for this were not well understood at the time. You can repeat this experiment by using a cycle dynamo and a telephone 'receiver' or a pair of headphones, but make sure to connect a 220Ω resistor in series to limit the current, and DON'T WEAR THE HEADPHONES, because the sound may be deafeningly loud. You can extend the experiment by equipping the receiver or one of the earphones with a conical horn made of rolled-up newspaper. This will make the sound much louder, and this effect has been known from very ancient times. As musical instruments, horns are said to have proved effective weapons of war at the Battle of Jericho (about 1500 BC), and the trumpets of Tutankhamen (1350 BC) have been played and recorded since their rediscovery in 1922.

The Telephone

The slowness of communication by telegraph, using manual Morse, was a considerable spur to the search for a practical telephone, which would convert the human voice into electricity and convert back again at the receiving end of the line. Many inventions were announced before Alexander Graham Bell produced a practical device in 1876. Much of the difficulty, in fact, was concerned with the design of a usable microphone. Both Ries in France and Hughes in England pursued the use of intermittent contacts, which could be made or broken by the vibrations of a diaphragm or other object exposed to the sound waves. Ries used metal contacts, which were unsatisfactory for the purpose but later proved useful as a detector of radio waves, in the 'coherer'. Hughes used carbon, which is very much more suitable, and produced many practical headphones.

Bell, however, used no contacts, his first microphones and receivers were identical and used electromagnetic induction, see Figure 1a. Bell's great contribution, apart from his commercial exploitation of the invention, was to realise that such a device, whether used as a microphone or a receiver, had to have a constant magnetic field, on which, in the receiver, the varying field due to the microphone current was superimposed. It can easily be seen that this constant field is necessary, because if the magnetic field acting on the diaphragm is produced only by the alternating microphone current, the diaphragm will be attracted to the magnet TWICE in each cycle of current, at both the positive and negative peaks. The receiver will thus produce sound at twice the frequency of the incoming current, i.e. there will be fullwave rectification and 67% second-harmonic distortion, see Figure 1b! A microphone with no constant field will just not work, because there is nothing to generate a current, but it is not so obvious that the continuous field in the receiver not only allows the correct reproduction of the input frequency but also dramatically increases the sensitivity.

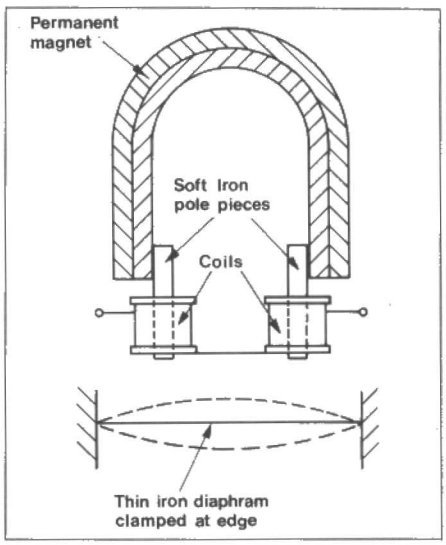


Figure 1a. Bell microphone or receiver with permanent magnet.

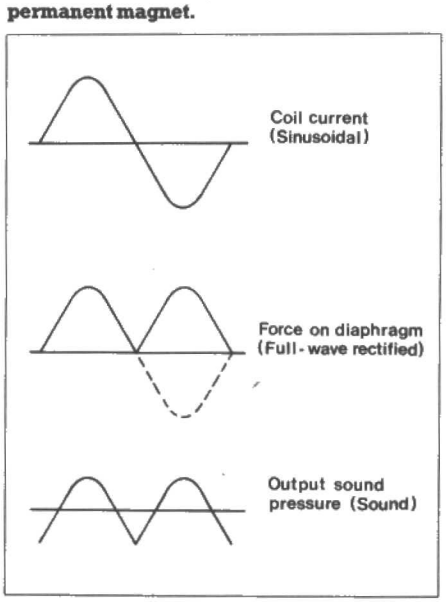


Figure 1b. Distortion produced by non-polarised receiver.

Bell used electromagnets in some early designs of transducer, but later he could make strong permanent magnets, and realised that the elimination of the battery power needed for the electromagnets was highly desirable. However, the best available material for making permanent magnets at that time was glass-hard high-carbon steel, and for the same energy-product, a magnet made of this material has to be far larger than even one made of cobalt steel, which was discovered by Honda and Takei in 1920, so that Bell's microphones and receivers were large and heavy. Their operating principle, however, is the same as that of the diaphragm-type earphone widely used between the World Wars, but the former were much more sensitive; they could be used in a 'sound-powered' system, where the devices were simply connected together, with no battery, and the energy for the system obtained from the sound input.

Later telephones used low-impedance microphones with batteries, and transformer matching to high-impedance receivers. Alternatively (and at that time all alternatives were investigated), a moving-coil receiver could be used with a low impedance microphone. This produced what Sir Oliver Lodge called the 'bellowing telephone', and could be said to be one of the first uses of a loudspeaker. It is used today as a door-answering device. Lodge himself patented a moving-coil loudspeaker in 1898. It must have been sufficiently different from the earlier Siemens device to be patentable. Another early application of the loudspeaker was the relaying of musical concerts to quite large audiences, particularly in France, using Bell-type transducers equipped with horns.

Problems with Bell-type Transducers

The sensitivity of Bell's devices is partly due to the design of their diaphragms. The diaphragm has to be made of ferromagnetic material, and should have a high permeability. But it should also be thin and springy, because it must be positioned very close to the pole-pieces of the magnet system, yet resist the attractive force of the permanent magnet. If the spacing is too small, or the diaphragm too slack, it will collapse in the middle and 'pole', or stick to the pole pieces, causing a great drop in sensitivity and considerable distortion of the sound.

For Bell, the magnetic and mechanical requirements were conflicting, since soft iron had the highest available permeability but not the ideal mechanical properties. Nevertheless, his diaphragms are described as 'soft iron', but are probably not of the purest low-carbon iron (Swedish iron). The thickness, temper and clamping of the edge were arranged, by accident or, most probably, design, to give a high-Q resonance at about 900Hz. This increased the effective sensitivity for speech very considerably. However, the small spacing of the diaphragm from the pole-pieces limited the permissible diaphragm movement (usually termed 'excursion'), and therefore the loudness of the sound produced when the device was used as a loudspeaker. In addition, the mid-band resonance gave an unpleasant coloured and strident sound quality to the reproduction of music, especially as the short, often conical, horn to which it was attached resonated in the same frequency range and would not reproduce lower frequencies properly (Photos 1a, 1b).

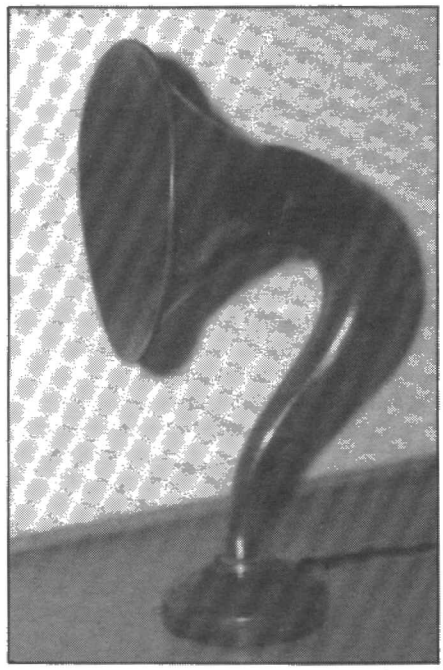


Photo 1a. Sterling 'Baby' horn with diaphragm transducer. About 1924.

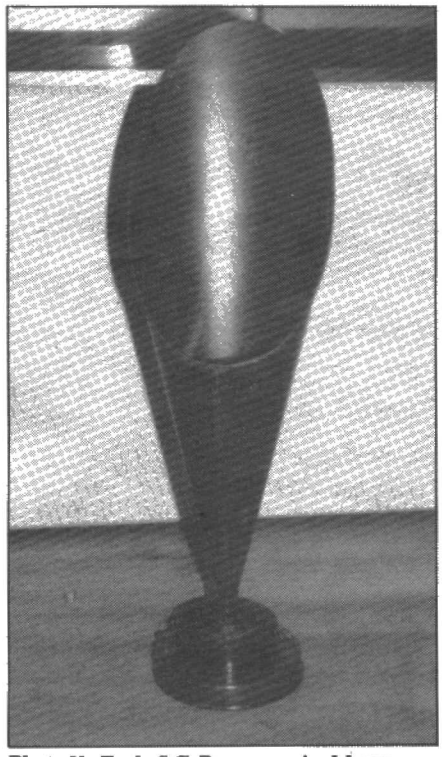


Photo 1b. Early S.G. Brown conical-horn loudspeaker.

The Cantilever Reed-armature Transducer

Based on a 1910 design for an earphone, by S.G. Brown, this device has the advantage of considerable simplicity and robustness, see Figure 2. Unlike the Bell receiver, it can drive a large, preferably conical, diaphragm directly, thus needing no horn (Photo 2a, 2b). Indeed, the mass of the diaphragm is highly desirable, as it lowers the resonant frequency of the reed. Millions of loudspeakers using this principle were made in the early days of broadcasting, and the driver mechanism was also used as a cutting-head for early electrical recordings on (analogue!) disc. Loudspeaker drivers were fitted with a screw adjustment of the spacing between the reed and the pole-pieces, which acts as a very effective volume control, much better than the proverbial sock which was stuffed down a horn to quieten it. However, too close adjustment of the reed results in poling, and the excursion is limited. In addition, as with the Bell transducer, the system is inherently non-linear. The pull on the reed increases when the reed moves towards the pole pieces, and decreases as it moves away, whereas it should be constant. To a certain extent this can be corrected by adjusting the geometry of the system, but it is possible to reduce the non-linearity by modifying the design more fundamentally, leading to the next two variants described below.

A loudspeaker with a large cone radiator can have its low-frequency response improved if a flat baffle-board or a cabinet (now usually called an enclosure) is added, see Figure 3, so that the sound radiated from one side of the cone is separated from the reverse-phase

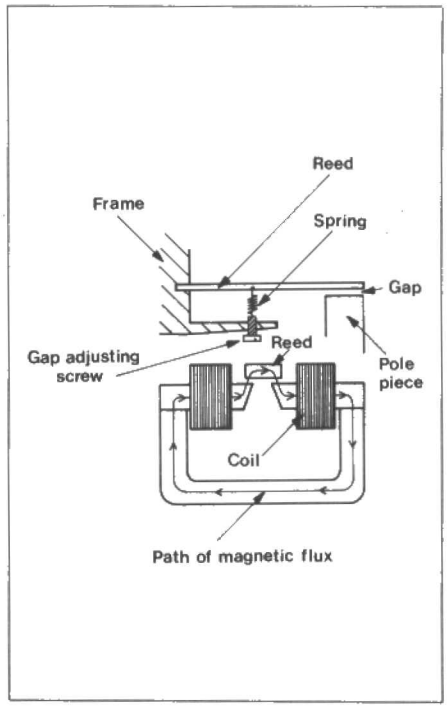


Figure 2. Reed armature driver.

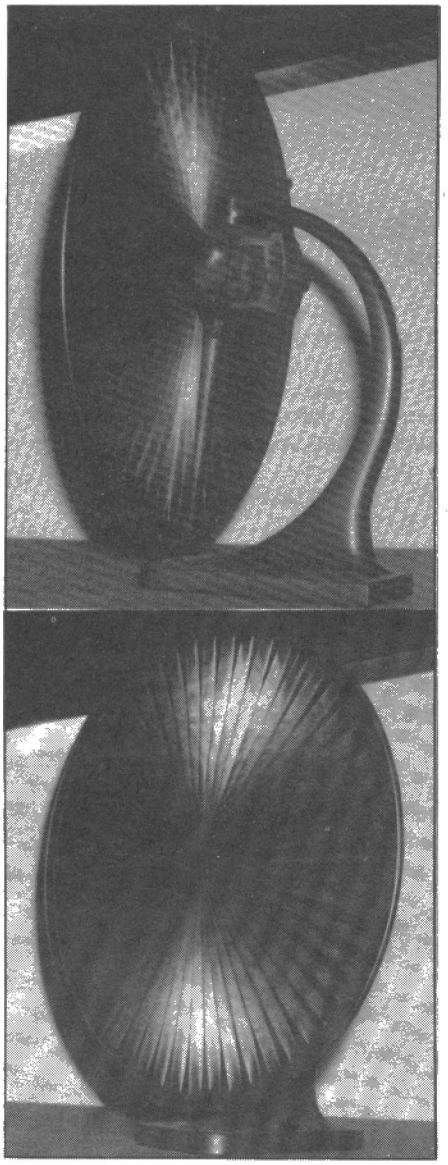


Photo 2a/b. Sterling 'Primax' pleated-diaphragm reed-driven loudspeaker. Contemporary sectioned exhibition model (1927). Many Sterling products were made under licence from The Gramophone Company Ltd., (later to become EMI).

radiation from the other side. While the effect of a very large flat baffle, and those of some other simple geometric shapes, such as a sphere, can be calculated, the effects of a rectangular box are quite complex, especially if the box is partly open at the back, as in the table radios and extension loudspeakers widely marketed in the mid-1920's and for the next forty years (Photo 3). Enclosure design has brought forth more oddities and weird theories than practically any other area of audio engineering. Successful theoretical analyses of the low-frequency response of some well-defined types of driver and enclosure combination were published by A.N. Thiele and R.H. Small in the 1970's.

The Inductor-dynamic Transducer

Developed specifically for loudspeaker applications, this device overcomes a major problem of the reed-armature mechanism, the inability to reproduce low frequencies unless a very low sensitivity is accepted. This is because the spring element of the reed has to be stiff enough to resist the steady pull due to the permanent magnet. This gives a high resonant frequency unless the combined mass of the reed and cone is made large, in which case the device requires a high power input, i.e. the sensitivity is low.

In the inductor-dynamic device (Figure 4), the direct pull and the alternating pull are arranged to be at right-angles, and the direct pull is, in addition, substantially balanced out. The residual direct pull is applied to the spring elements, of which there are two, in such a direction as to stretch them, whereas the alternating pull bends them.

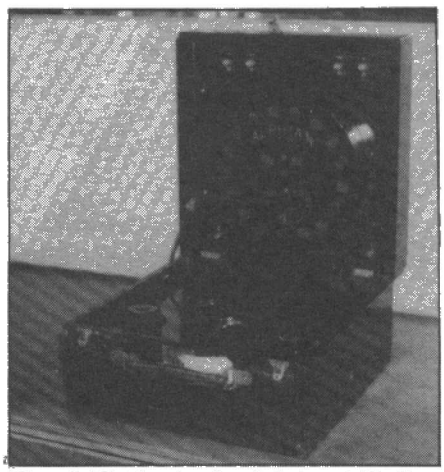


Photo 3. Alphion 4-valve (tube) portable (!) radio with Celestion reed loudspeaker (about

A simple flat spring is very much more difficult to stretch than to bend, so bending can be made easy without risk of poling. Also, the non-linearity is approximately balanced out, if the pole-piece tips are carefully shaped, because the reduction of force on one armature as it moves away from its pole-pieces is compensated by an increasing force on the other armature. This is an example of a push-pull mechanical system, which, like its electrical analogue, is substantially free of even-order non-linearity. With soft springs and a 25cm paper cone having a leather surround at its outer edge, a resonant frequency in the region of 70Hz could be obtained, whereas the resonance of a reed-armature loudspeaker might be above 200Hz. In addition, the inductor-dynamic driver cannot pole, so the amplitude of movement is limited only by the tolerable non-linearity, and can be increased by appropriate armature and pole-tip design. Thus the quality of music reproduction from an inductor-dynamic loudspeaker could be much better than that from a reed-armature type.

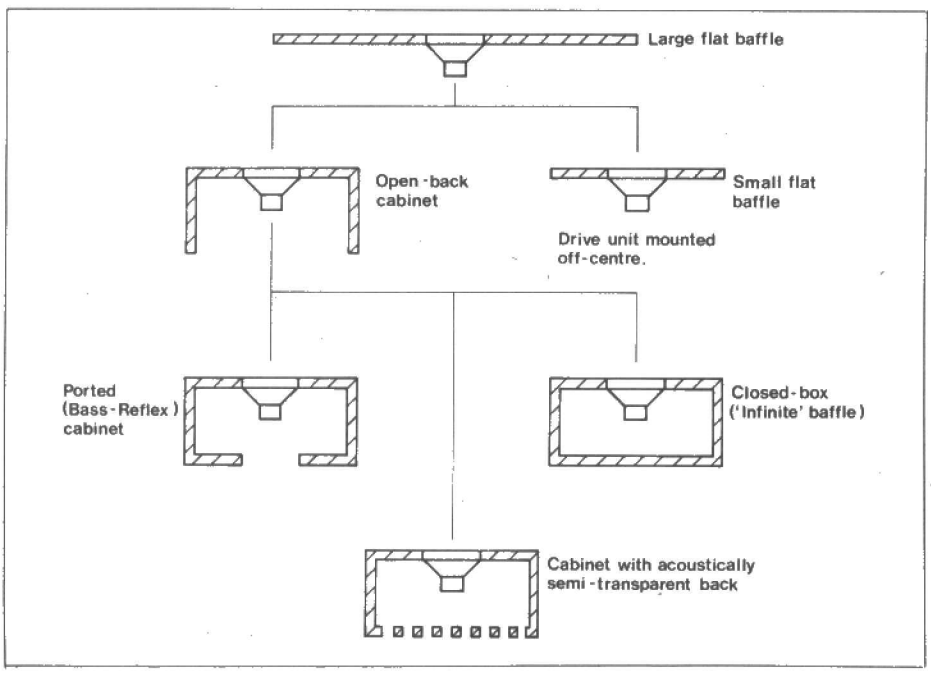


Figure 3. Some types of baffle for direct-radiator loudspeakers (modified and extended from Jordan).

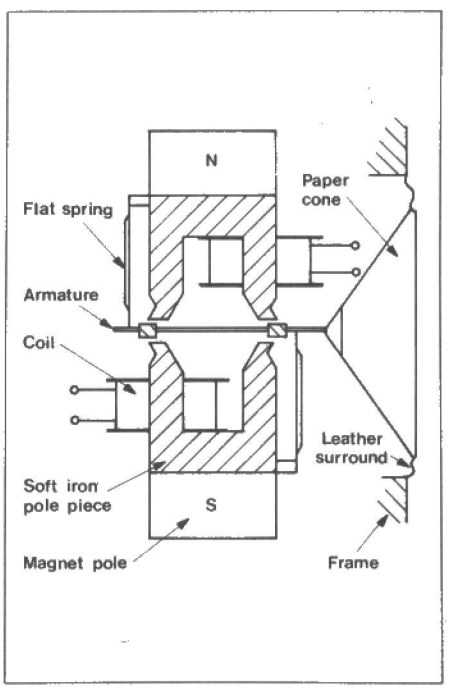


Figure 4. Inductor-dynamic loudspeaker.

The Balanced-armature Driver

This is another way of improving the linearity of the net driving force on the armature system. With the rocking armature (Figure 5) in the exact centre position, there is no net direct pull on it, but it is unstable in this position, like a horizontal see-saw. Consequently, a spring has to be added to prevent poling. This is a pity, because the effective mass of the armature is quite small, so the spring results in a high resonant frequency unless a heavy cone is added. Only the alternating component of the magnetic field passes through the armature, which is therefore not driven close to saturation by the permanent field. The permeability of the armature can therefore be high, with a consequent high sensitivity. This is another push-pull system, and linearity can be similar to that of the inductor-dynamic device. The balanced-armature mechanism may be rather easier to construct, although play in the armature pivot must be prevented.

Electrostatic Loudspeakers

The electrostatic or capacitor microphone predates Bell's telephone by some years, but, because of its high impedance, it could not form part of a practical telephone at that time. Indeed, it is only now, a century later, that this is practicable. The early microphone could also be used as a loudspeaker but required high operating voltages (upwards of 600V) and was not very sensitive. It was not until the end of the 1920's that new materials, such as aluminium foil and thermosetting plastics (Bakelite), became available and new patents for 'condenser loudspeakers' began to appear, in the names of V.F. Greaves *et al.*, C. Kyle, P.E. Edelman and H. Vogt. Greaves, Kyle and their colleagues produced a single-sided unit, see Figure 6a, while Vogt produced a unit with two perforated fixed plates enclosing a stretched moving plate, see Figure 6b. This unit therefore resembled the much later and considerably more successful 'Quad Electrostatic', but did not share the latter's crucial constant-charge drive, and was therefore not very linear in amplitude response, although the frequency response was said to be fairly smooth. The Kyle unit was also said to have a frequency response extending from 100Hz to 10kHz, which was not flat but could be equalised fairly easily. Being single-sided, however, the linearity of the device would be suspect.

Piezo-electric Loudspeakers

The term 'piezo-electric' refers to the property of certain materials to deform mechanically when an electric field is applied, and vice versa to generate a voltage when mechanically stressed. In the early 1930's, loudspeakers, mostly for reproducing frequencies above 5kHz, were described by S. Ballantine, C.B. Sawyer and F. Willms. These used crystals of Rochelle salt (sodium potassium dihydrogen tartrate) as the piezo-electric material. This is very difficult to work with, because it attracts moisture from the air and dissolves in it. Quartz or tourmaline would be much more stable but are much too insensitive for use in loudspeakers. The problems of sealing the crystals against moisture were not solved and the principle was abandoned until the 1950's, when it was briefly revived. New ceramic materials developed since 1970 have allowed the production of reliable piezo-electric high-frequency drivers, but the extension to lower frequencies is difficult. Linearity can be poor, and the capacitive input impedance of the device is not easy to drive.

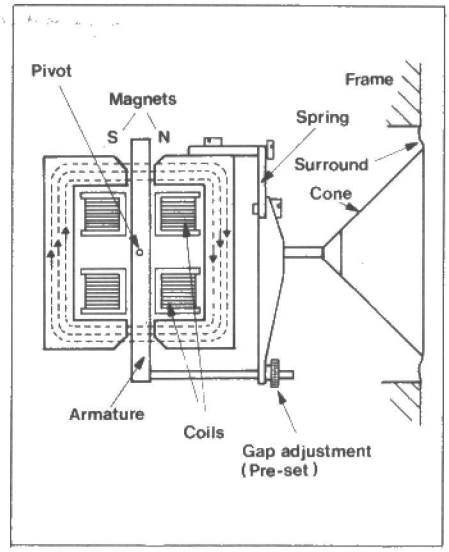


Figure 5. Balanced-armature loudspeaker. Dotted lines show magnetic flux path with no coil current. There is no flux in the armature.

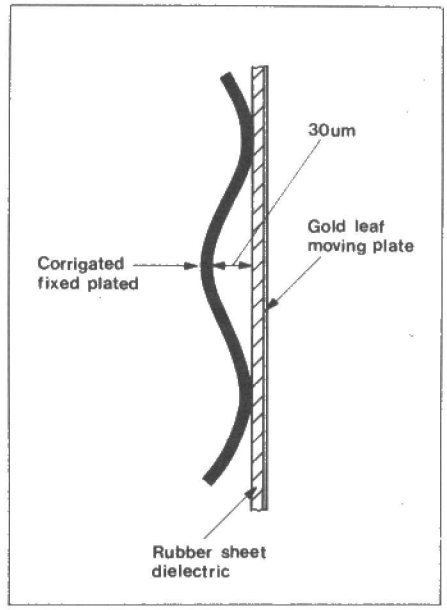


Figure 6a. Modified cross-section of Kyle single-sided electrostatic loudspeaker.

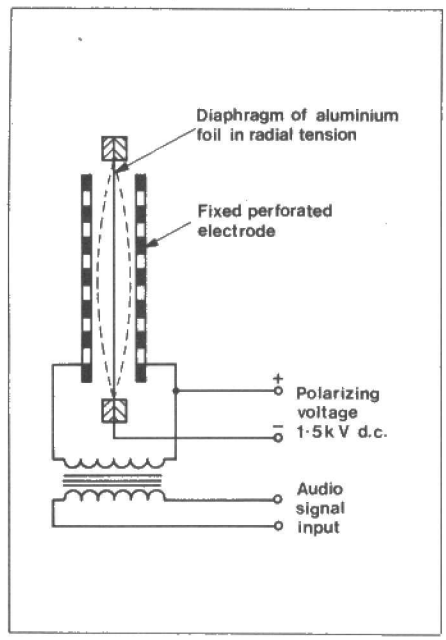


Figure 6b. Cross-section of Vogt double-sided electrostatic loudspeaker.

Friction-driven Loudspeakers

In attempts to produce very sensitive loudspeakers, and/or very high sound levels, electromechanical devices were produced in which the friction between a pad, connected to a conical diaphragm, and a rotating disc or roller was varied electrically. This could be done either by varying directly the force between the pad and the moving part, or by impregnating the moving part (a porous cylinder) with a mixture of chemicals which gave off gas bubbles when electrolysed by the applied electrical signal. Such devices were described by Thomas Alva Edison, Johnson Rahbek and S.G. Brown, but were not very reliable and did not give very good or consistent sound quality. The signal-to-noise ratio was also poor, because the friction introduced noise in the absence of an input signal.

Air-operated Loudspeakers

Another high-power system, developed by Creed and Co. in the 1930's, used a supply of compressed air, the flow of which was controlled by an electrically-operated vane valve. Apart from distortion due to deficiencies in the valve operation, there was considerable non-linearity due to the sound pressure variations at the horn throat being far from negligible compared with atmospheric pressure. This problem also occurs with modern high-power horn loudspeakers, which use moving-coil drivers.

The Moving-coil Driver and Direct-radiator Loudspeaker

By far the most widely used type of driver, the moving-coil system (Figure 7) is surprisingly subtle in its mode of

operation, and it is this which confused the early workers and was partly responsible for the long delay in its exploitation. The other factor is that the construction of a reliable device depends on the use of stable flexible materials and adhesives, neither of which were easily obtained before the 1920's.

The moving coil and cone assembly can be made very light. If provided with a very soft suspension (Photo 4a, 4b and 4c), the result may be a very large coil excursion under some conditions, causing the destruction of the coil by collision with the magnet structure, or by being forced out of the magnet system. On the other hand, a stiff suspension will result in a very high-Q resonance, giving a squawky or strident sound, and low sensitivity except at the resonant frequency. The major contribution of Rice and Kellogg was not the breakfast cereal but the realisation of how to make a moving-coil loudspeaker with a substantially flat frequency response. This involves three steps:

(a) Adjusting the mass of the coil and cone, and the compliance of the suspension, so that they resonate below, or at the lower end of, the working frequency range.

(b) Adjusting the amount of mechanical loss (mainly in the suspension and surround), and the electrical damping, which depends on the output impedance of the driving amplifier and the field strength of the magnet system, so that the main resonance is reasonably well-damped. (Rice and Kellogg may not have fully analysed this: Theile and Small provided a full explanation and quantified the factors involved some 25 years later.)

(c) Allowing, or even encouraging, the cone to cease to vibrate in one piece above a certain frequency, but not to vibrate in sections in several undesirable ways.

Curiously, Siemens' original patent refers to a conical diphragm with an exponential flare: this shape encourages the correct form of cone break-up and has been used in some of the nicest-sounding commercial (as opposed to high-fidelity, where such simple criteria are inadequate) loudspeakers.

In addition, Rice and Kellogg showed that the diaphragm should be small compared with the wavelength of the sound to be radiated (unless a pronounced directional effect is required), and indicated the need for a baffle or enclosure to prevent destructive interference between front and rear radiation at low frequencies.

Until the late 1930's, permanent magnets were not very suitable for use in moving-coil drivers. The present author remembers dismembering an early example, from a battery radio, in his youth: the magnet was forged from about 30cm of 75mm×25mm steel bar, rolled into a flattened hoop, welded and fully hardened. It was very heavy indeed, yet not a very strong magnet. The earlier Rice-Kellogg products, made by GE in

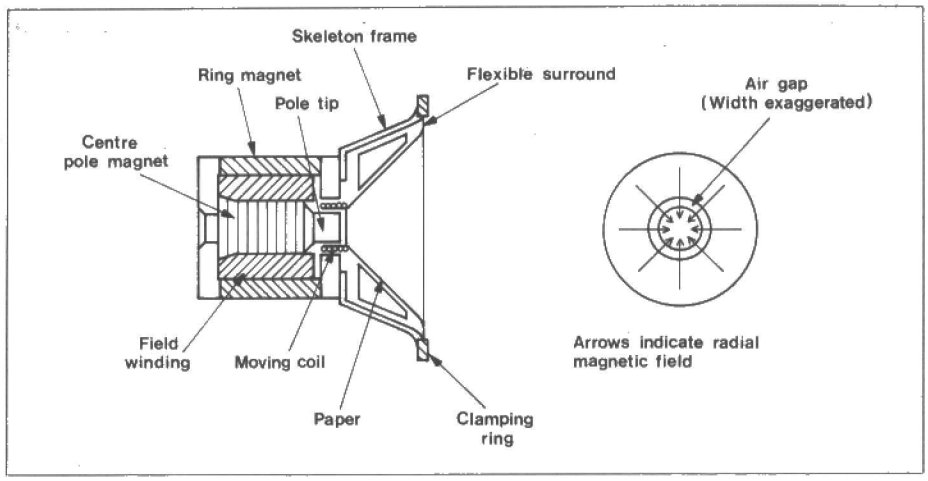


Figure 7. Moving-coil loudspeaker with metal magnet or electromagnet. The two permanent magnets and the field coil are alternatives: whichever magnet is not fitted is replaced by soft iron.

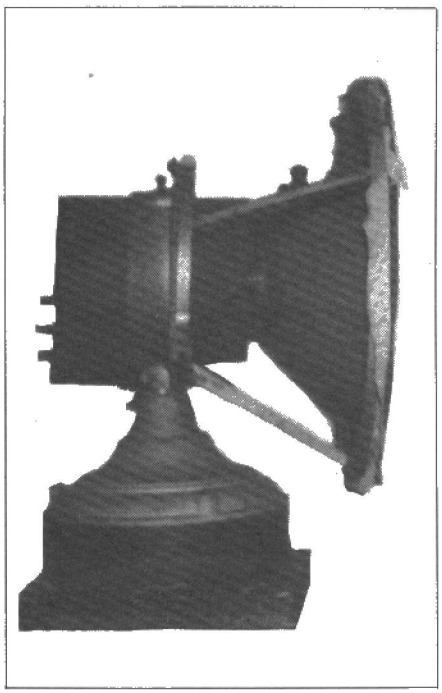


Photo 4a.

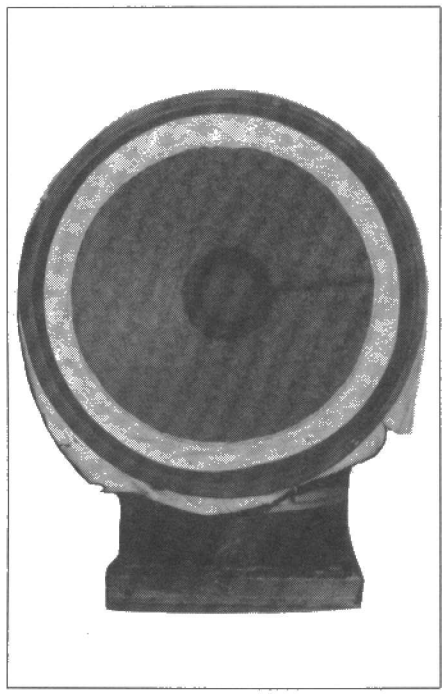


Photo 4b.

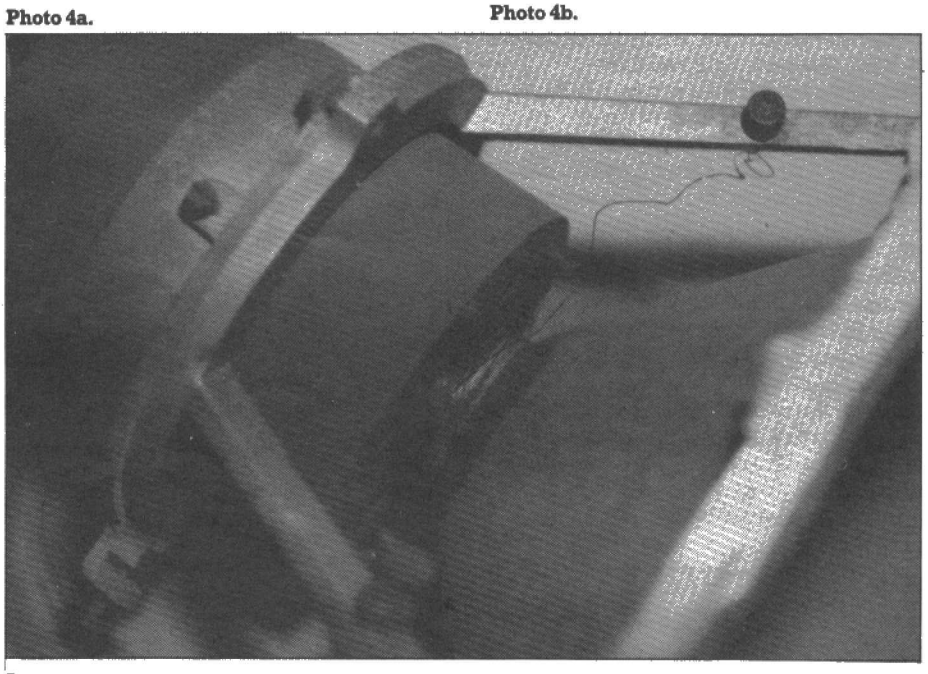


Photo 4a, b, c. Bakers Selhurst Radio 12" moving-coil direct-radiator loudspeaker with a high-impedance voice-coil. This unit has a leather surround and the suspension is formed by two loops of sewing-thread. The resonant frequency is very low, but so is the maximum permissable input.

the USA and marketed in Britain by BTH (Photos 5a, 5b, 5c and 5d), used electromagnets, as did most of the units made at that time. When used in a radio receiver, the field coil served as the smoothing inductor in the power supply. Considerable current at harmonics of the mains supply frequency also flowed through the coil: the reservoir capacitor was only 4μF or 8μF, and the smoothing capacitor, following the inductor, was of the same value. This current would have produced a loud hum from the loudspeaker, so another coil of a few tens of turns was wound next to the field coil and connected in series with the voice-coil. This 'hum-bucking' coil inserted a voltage into the voice-coil circuit which was intended to cancel the effect of the hum current in the field. It did work, but the coil had to be specially designed for each radio, since the number of turns depended on the power supply capacitor values, the anode current drawn by the valves (tubes) and the output source impedance of the output stage.

Flat-diaphragm Moving-coil Loudspeakers

There were some researchers who did not like cones: certainly they are far from ideal, but all other practical shapes have their own disadvantages, which seem to be worse than those of the cone. One of the proponents of the flat diaphragm was J.D. Midgley, who patented several arrangements, including the use of a circular, stretched aluminium foil diaphragm which was driven by a moving coil mounted off-centre (Photo 6a, 6b). The off-centre drive helped to break up the inevitable resonances and spread them out in the frequency domain. Another type of flat radiator was the German Blatthaller (sheet sounder), in which a large corrugated metal sheet (up to several tens of centimetres square) was driven all over its surface by a copper conductor fixed to it at right angles and immersed in the field of a powerful electromagnet. It was very large, very heavy, and very loud.

Horn-loaded Moving-coil Loudspeakers

Moving-coil drive units were not only used in direct radiators. The talking cinema and the growing sound-reinforcement industry both demanded high sound levels and high quality. It was relatively more difficult in those days to provide more amplifier power than to improve the sensitivity of the loudspeakers. E.C. Wente and A.L. Thuras (the inventor of the bass-reflex principle) described in 1928 the Western Electric WE555 horn-loaded moving-coil unit. The coil was made of edge-wound aluminium tape, and, whereas most loudspeakers are of the order of 1% efficient (sound power out divided by electrical power in), this unit was about 50% efficient! Modern horn units approach the same performance (with

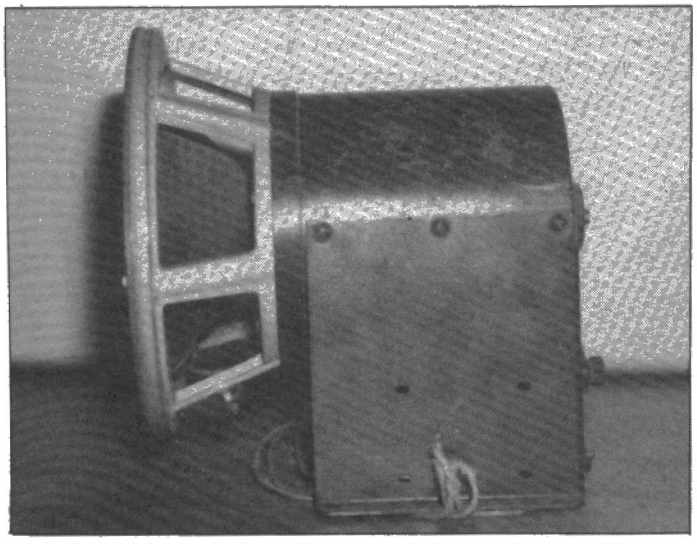


Photo 5a. B.T.H. Rice-Kellogg 12" energised-field loudspeaker 1931.

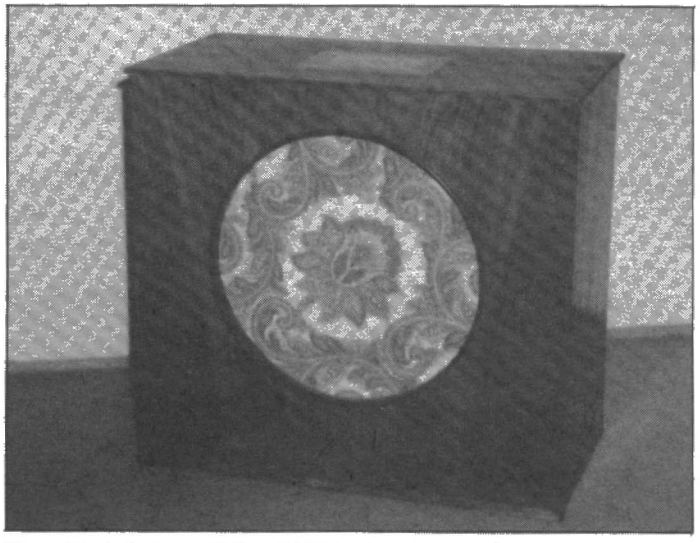


Photo 5b. A similar unit from 1929, in the original cabinet with its field power-supply.

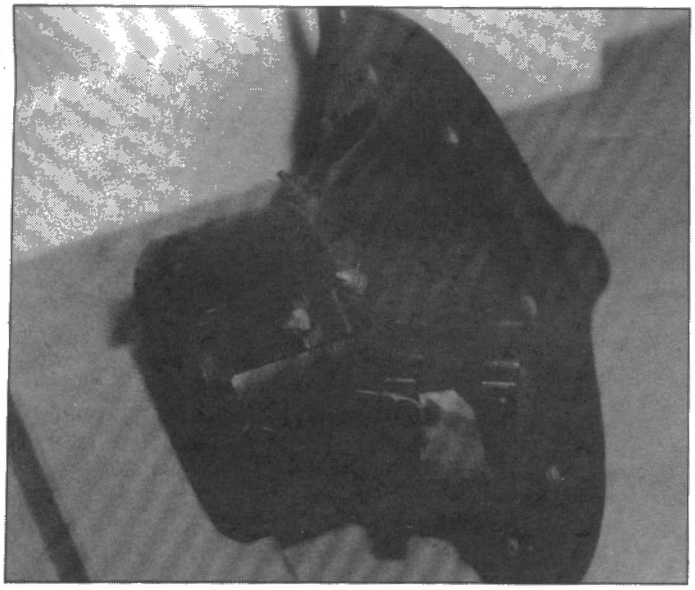


Photo 5c. A 12" permanent-magnet unit from the same stable (1931). The slot, one of three in the magnet pot, is not for displaying the interior but a feature of the design.

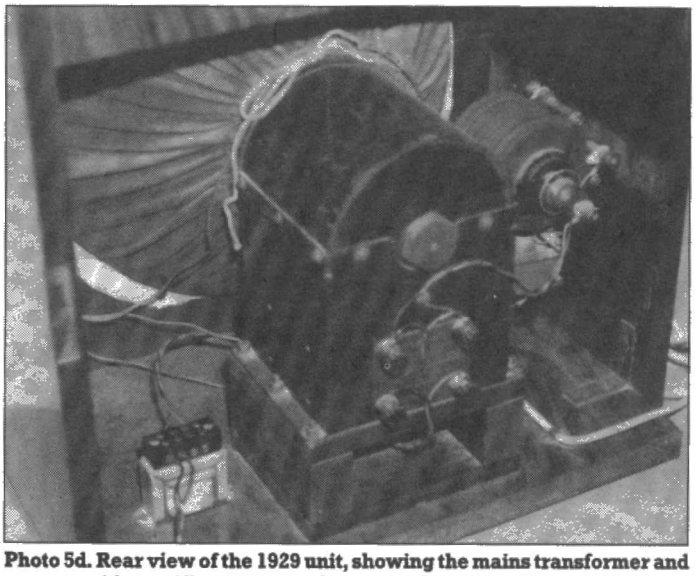
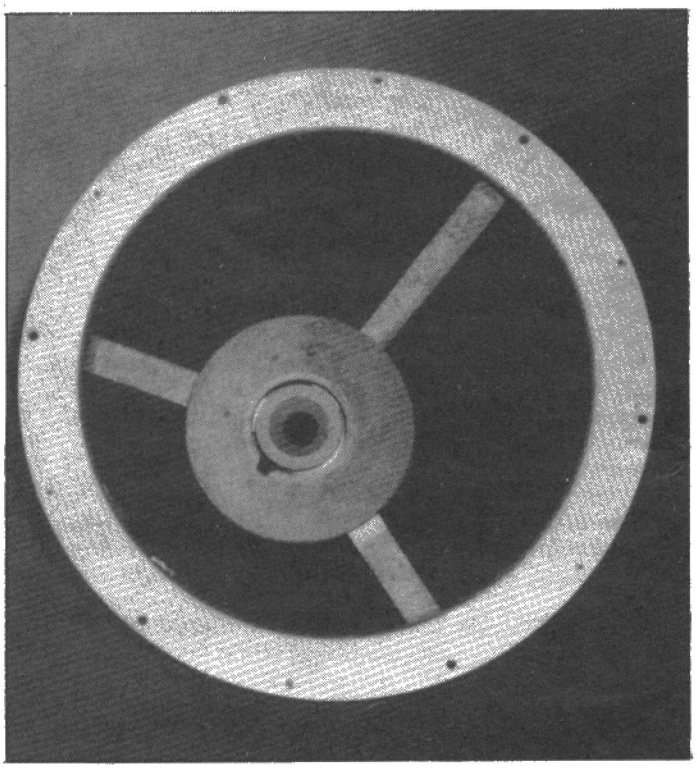


Photo 5d. Rear view of the 1929 unit, showing the mains transformer and copper-oxide rectifier. The transformer on the left is a Wharfedale WMT 1, for 3Ω:15Ω matching. Many thousands of these were made, from the 1930's until relatively recently. This example is a late model, for it is branded 'Rank-Wharfedale'.



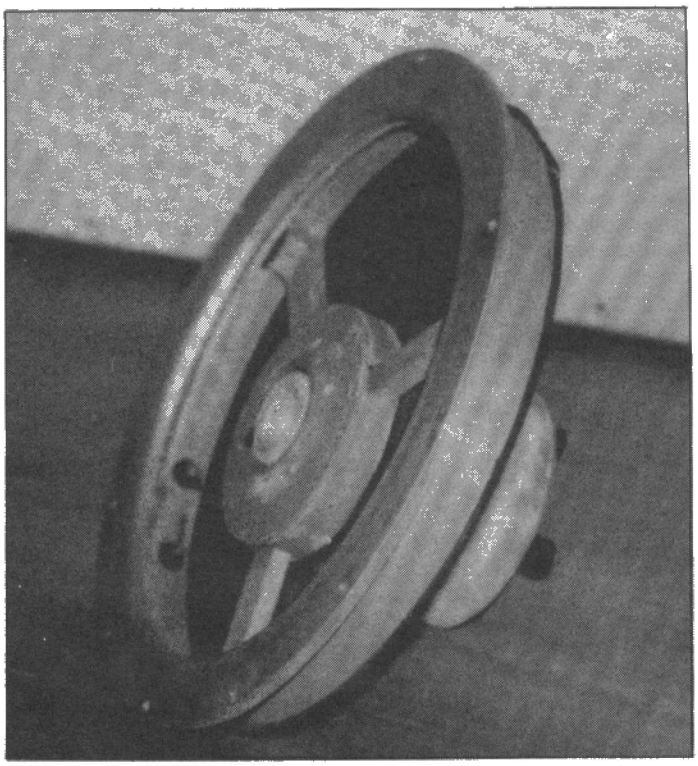


Photo 6a, b. Two views of the chassis of a Midgley flat-diaphragm moving-coil dipole-radiator loudspeaker. Note the off-centre drive, intended to minimise spurious resonances in the thin aluminium diaphragm (absent from this example).

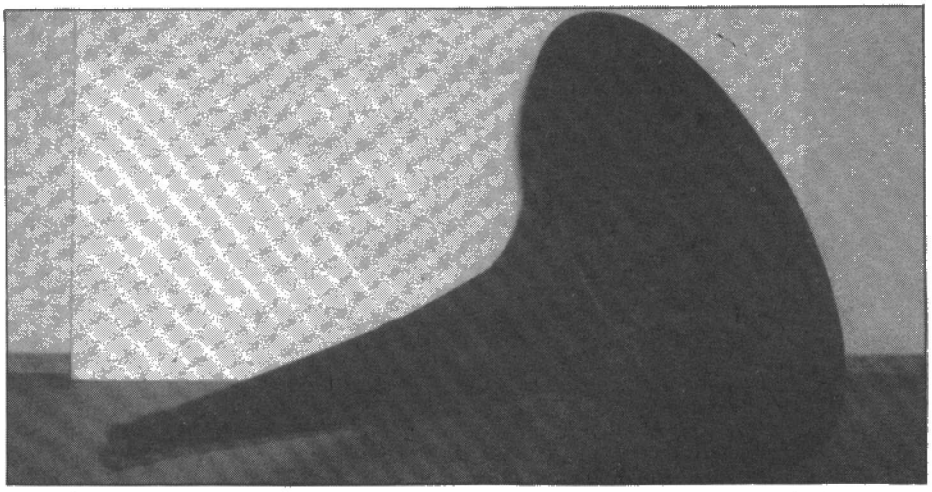


Photo 7. A curved-axis exponential horn, from a Magnavox horn-loaded moving unit of uncertain (but very early) date.

less size and mass and less relative cost) but few indeed exceed it. Mind you, the horn was 10.8 m long! A cross-section of a horn-loaded moving-coil pressure unit is shown in Figure 8. Photo 7 shows a horn from a Magnavox horn-loaded moving-coil unit.

One of the more unusual devices of the day was the 'Crystavox' loudspeaker designed by S.G. Brown, which incorporated a 'microphonic amplifier' (a term with quite a different meaning today!). In photos 8a and 8b, the unit in front of the base of the horn is the amplifier, with its metal cover removed. At the left of the assembly is a reed mechanism which is directly mechanically-coupled to a small carbon microphone cell on the right. This is

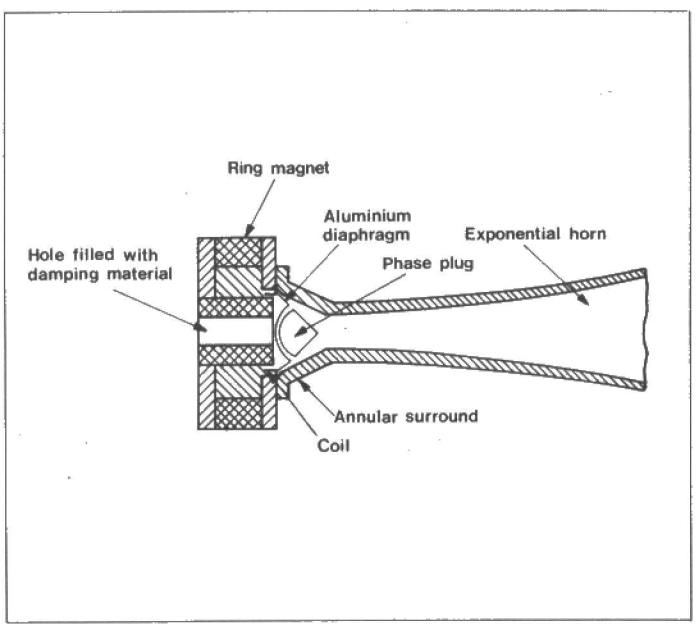


Figure 8. Horn-loaded moving-coil loudspeaker. Early examples used an electromagnet field.

Photo 8a. The 'Crystavox' incorporating a 'microphone amplifier'.

connected in series with an external 3V battery and the low-impedance reed mechanism at the base of the horn. It worked well, and the principle was known and used experimentally before 1900 and certainly long before the triode valve was available as an amplifier.

The End of the Beginning

The start of World War II in 1939 could be regarded as the end of the first stage of the loudspeaker story. Loudspeakers were being produced by the million for mass markets. New magnet materials, iron alloys containing cobalt, nickel and aluminium, and new methods of heat-treatment, were being introduced which would reduce size, weight and, in spite of the exotic metals used in the alloys, cost. The performance of a loudspeaker design could, to a certain extent, be predicted, once the characteristics of the cone to be used had been measured. Cone design was to remain a black art for another thirty years or so, and unpleasant (and even, occasionally, pleasant) surprises still lie in wait for the innocent designer.

Acknowledgement

The assistance of Mr. G.M. Wheeldon of the Science Museum and Ms. J. Marshal of the Science Museum Library is gratefully acknowledged.

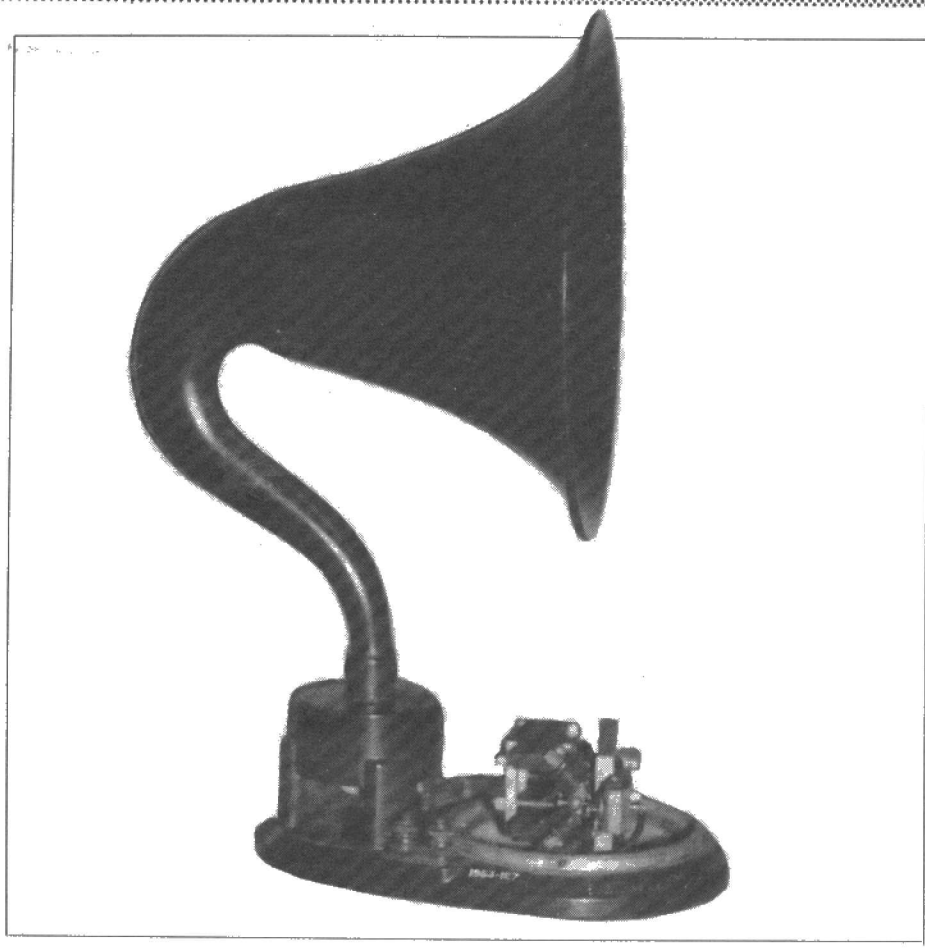
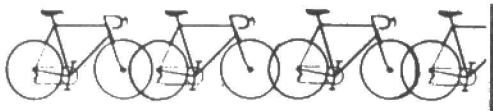


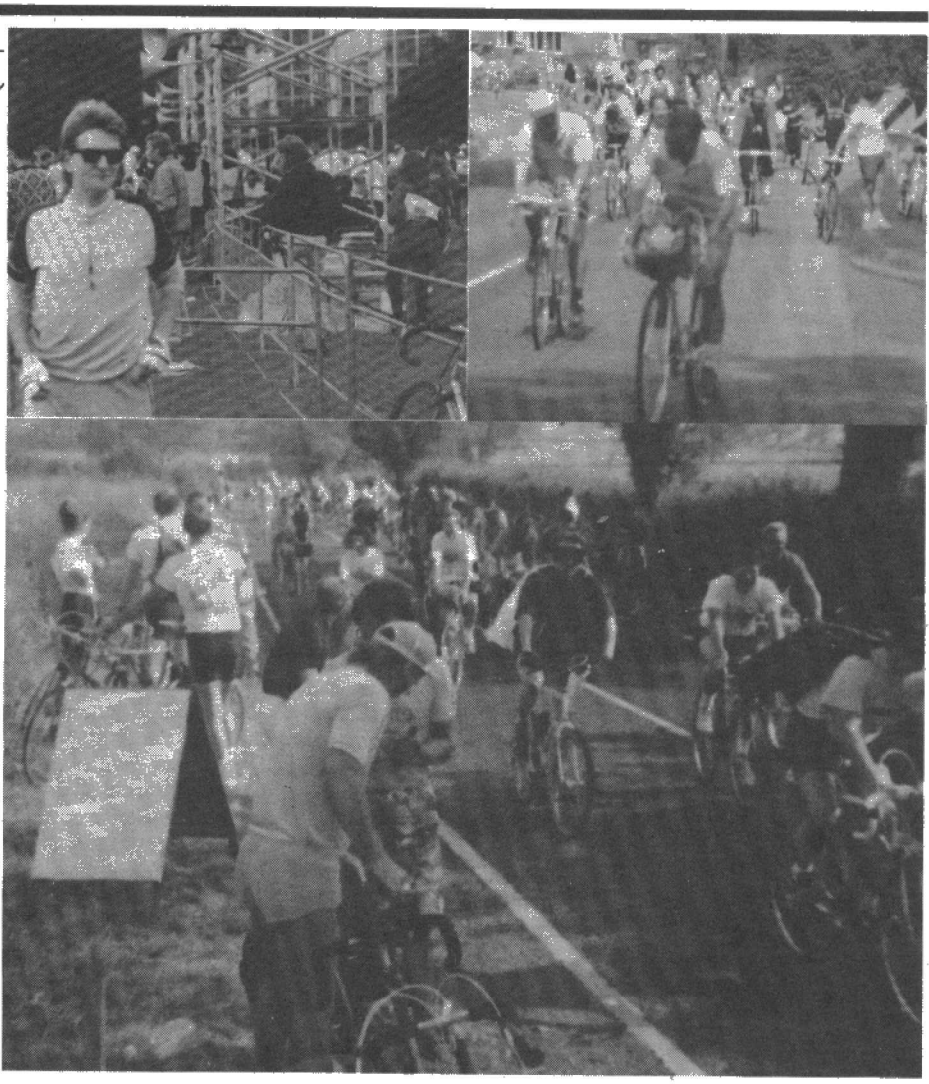
Photo 8b. Another view of the 'Crystavox'.



Maplin's Big Heart

By J. Rose

In the last issue I told you about the annual London to Brighton bicycle ride in aid of the British Heart Foundation, this ride has now taken place and was thoroughly enjoyed by all! Over 35,000 cyclists suffered the 56 mile journey with smiles on their faces, joy in their hearts, blisters on their feet and saddle sores on their ....! The event raised an estimated £1,000,000 which will be used to fund the purchase of much needed equipment and be used for invaluable research work. Maplin contributed over £200 to the Foundation and I would like to thank the staff of the Hammersmith and Birmingham shops particularly and those customers who came forward and donated. Money was also raised at the Head Office in Hadleigh, Essex and my gratitude goes to Hazel and Beryl for their gentle bullying! It is hoped that this will become a Maplin tradition with more support building up each year, as this is a very worthy cause and, let's face it, it only costs Dave Kirk and myself a few aches and pains, not to mention those saddle sores!



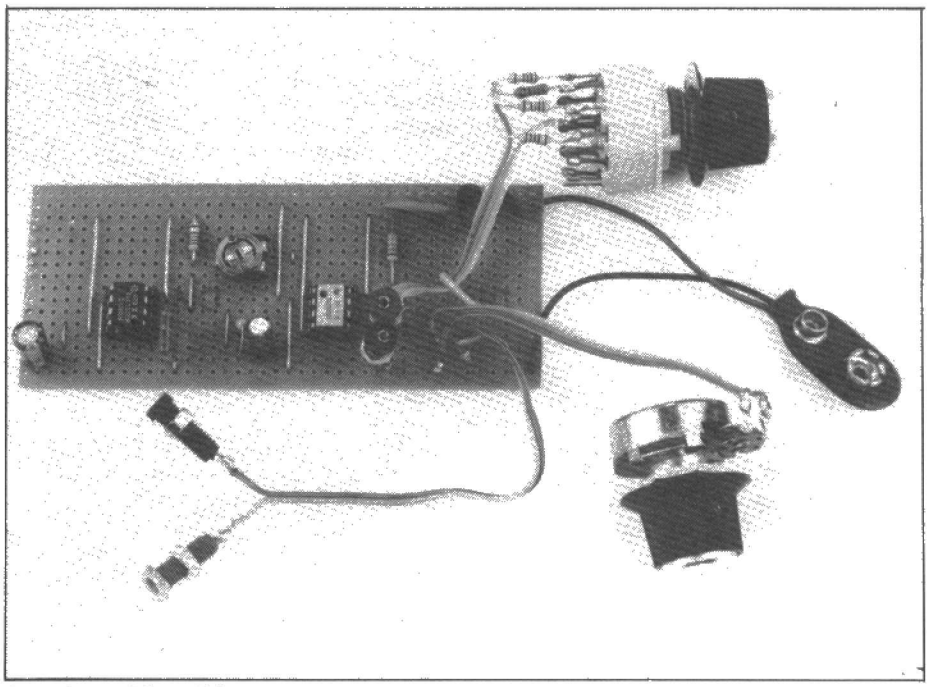
Bob's MINI-CIRCUITS

From Robert Penfold

Capacitance Meter Adaptor

A capacitance meter is certainly one of the more useful items of equipment for the home constructor, but even home built instruments tend to be rather expensive. Unfortunately, few multimeters seem to include any capacitance measuring ranges. A low cost but very effective method of capacitance measurement is to use an add-on circuit for an existing multimeter. This avoids the main expense incurred when building a stand-alone unit, which is the cost of the digital display circuit or a good quality panel meter. The capacitance meter add-on described here is primarily intended for operation with a digital multimeter switched to its 1.999 volt range, but it should work quite well with an analogue multimeter having a sensitivity of 20k/volt or more, and a D.C. voltage range of 1 to 2.5 volts full scale. It has four ranges with full scale values (when used with a digital multimeter) of 1.999nF, 19.99nF, 199.9nF, and 1.999μF full scale. High value capacitors are beyond the capabilities of this circuit, but its range is quite respectable for such a simple and inexpensive circuit. Anyway, high value capacitors can usually be given a rough check with a multimeter set to a resistance range. The charge current of the capacitor (or lack of it) will be indicated by the meter, and will show whether or not it is basically functional. At the low end of its range, this unit can check capacitors of values down to a few picofarads.

The circuit is based on two 555 timers (IC1 and IC2) which are used in the astable and monostable modes respectively. IC1 provides a clock signal which is a series of very brief negative pulses. These trigger IC2 which provides output pulses of a duration that is controlled by the values of the switched resistors (R2 to R5) and the capacitor under test. The switched resistors provide the unit with its four measuring ranges. With the frequency of the output signal at a fixed rate, the average output voltage of the monostable depends on the duration of the output pulses, and is proportional to this duration. This is in turn proportional to the value of the test capacitor. The circuit therefore provides the required capacitance to voltage conversion. RV1 is used to calibrate the unit. R2 to R5 have a tolerance of 1% so that consistent results are obtained from one range to the next. Most multimeters have internal smoothing components, but R6 and C3 will smooth the output to a low ripple D.C. signal in the absence of any built-in smoothing.



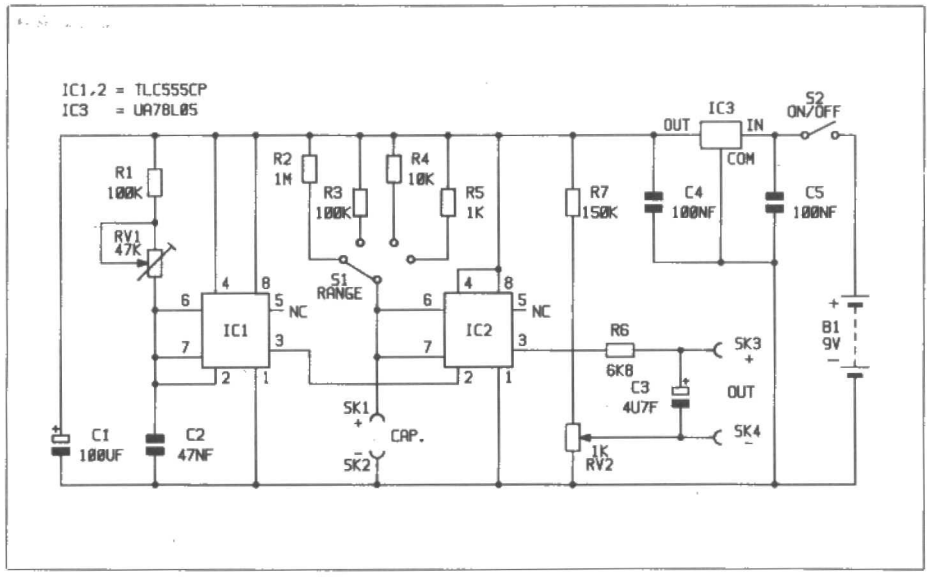
Capacitance Meter Adaptor.

There is a problem with simple capacitance measuring circuits of this type in that there is a certain amount of stray capacitance in the monostable which is in parallel with the test capacitor, and adversely affects accuracy. The problem is insignificant on the higher ranges, but can be quite severe on the lower ranges. In this case the problem is eased by the use of the TLC555CP for IC1 and IC2. This has a higher switching speed and lower internal capacitance than the standard 555, and a quiescent reading of only about 20pF is obtained. The problem is further eased by RV2, which provides a variable offset voltage that can be used to zero the quiescent reading. This enables the unit to measure very low value capacitors.

A stable 5 volt supply is needed, and this is obtained from a 9 volt battery via monolithic voltage regulator IC3. The current consumption of the circuit is only about 4 milliamps.

Reasonable care must be taken with the layout of any circuit of this type, so that the stray capacitance in parallel with test components is kept as low as possible. I originally mounted R2 to R5 on the circuit board, but this boosted the stray capacitance from about 20pF to 80pF. It is much better to mount these resistors on S1. The unit can be calibrated on any range, but for best accuracy use a close tolerance capacitor that has a value which is around 50% to 100% of the full scale value of the

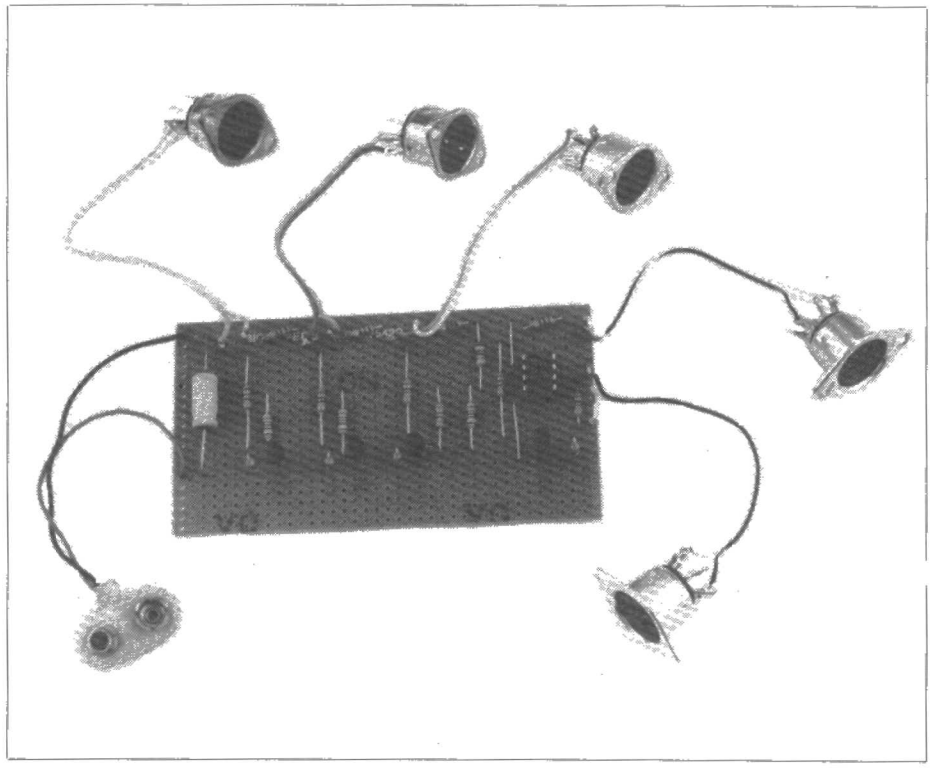
range on which the unit will be calibrated. Zero the unit using RV2 prior to connecting the test component, and then adjust RV1 for the correct reading on the multimeter. Note that changing ranges will require some slight readjustment of RV2 in order to re-zero the unit. Note also that the decimal point of the display is in the wrong position on the middle two ranges, but you will soon get used to converting readings into true capacitance values. The unit is quite accurate, but when testing capacitors you must remember to make allowances for small deficiencies in the accuracy of the unit, and the tolerances of the test components (which can often be some 20%, or even more for ceramic types). As with any capacitance meter, do not connect charged capacitors to it! Always discharge test capacitors before connecting them to the unit.



Capacitance Meter Adaptor circuit.

MIDI Thru Box

The standard method of driving pieces of MIDI equipment from a control device is the so-called 'chain' system of connection. This has the 'OUT' socket of the controller connected to the 'IN' socket on one of the other pieces of equipment, and then the 'THRU' socket of this device connects to the 'IN' socket of the next piece of equipment, and so on. In theory, any number of instruments, etc. can be connected together by wiring the 'THRU' socket of one instrument to the 'IN' socket of the next one in the 'chain'. In practice this is not always possible though. There can be problems with what are often called 'delays', but which are more probably problems with smearing of the signal that compromise reliability. At a more basic level, many items of MIDI equipment (especially keyboard instruments) simply do not have a MIDI 'THRU' port, and cannot be used with this method of connection. Actually, if only one instrument lacks a 'THRU' socket it is possible to use the 'chain' system, provided this instrument is placed at the end of the 'chain'. If more than one instrument lacks a 'THRU' port, then the 'star' system must be used.



MIDI 'thro' stripboard layout.

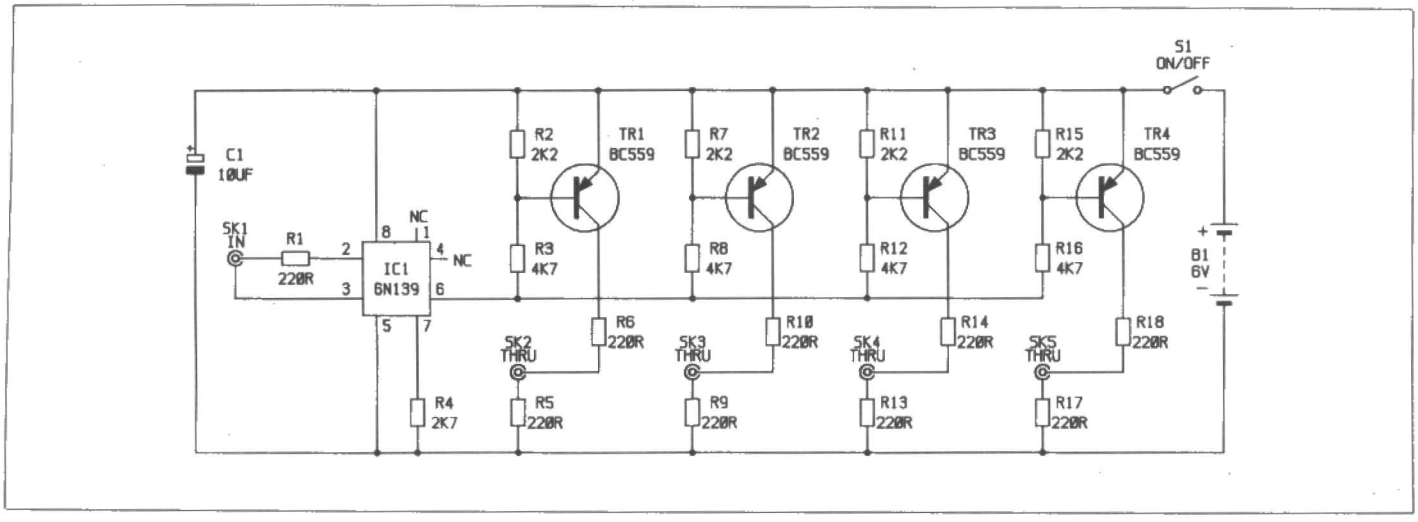


Figure 1. The MIDI 'THRU' circuit.

It is only possible to adopt the 'star' system if the MIDI controller has multiple outputs, or a 'THRU' box is included in the system. This method of connection relies on the controlling device having an 'OUT' socket for each MIDI input that must be driven. A 'THRU' box simply takes the signal from a MIDI output and splits it to give a number of 'THRU' outputs to drive the inputs of the other equipment in the system. A 'THRU' box cannot be a passive device as a MIDI output only provides a drive current of 5 milliamps, and splitting this between several inputs would give an insufficient drive current for each one.

In this circuit, shown in Figure 1, an opto-isolator is used at the input. This is not strictly necessary as there is no need for a 'THRU' box to provide isolation, but MIDI outputs are designed to drive an opto-isolator, and this method ensures reliable operation with any output that properly meets the MIDI standard. The 6N139 used in the IC1 position is not a simple LED/transistor type, but on the output side actually has a photo-diode, an emitter follower transistor, and a common emitter output stage. This gives high efficiency and fast operating speed. The circuit can comfortably accommodate MIDI's fairly high baud rate of 31250 baud, R4 ensures that the emitter follower stage operates at a reasonable current and that the device achieves a suitable fast switching speed. The 5 milliamp drive current is set partly by R1, and partly by a series resistor in the drive circuit.

On the output side of the circuit there are four common emitter switching transistors, with each one driving a separate 'THRU' socket. Two current limiting resistors are used in each output circuit, and this two resistor system gives better protection to the circuit in the event of a system being incorrectly wired up, or a fault occurring. Four output stages are shown in the circuit diagram, but IC1 is capable of driving several more output stages if necessary. Power is provided by a 6 volt battery (such as four HP7 size cells in a plastic holder). The quiescent current consumption will probably be negligible, but under worse case conditions the average current drain could be as much as 2.5 milliamps per output that is actually used.

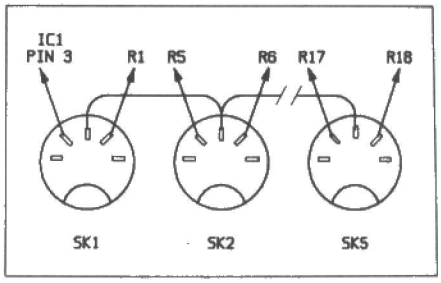


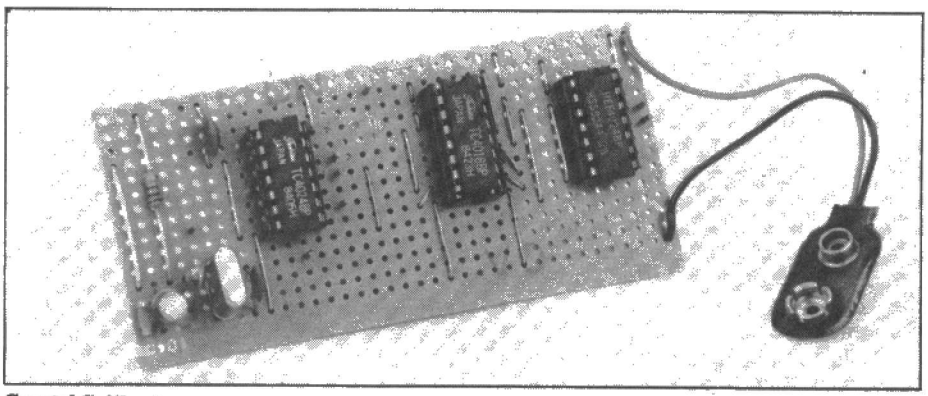
Figure 2. Socket connections.

The standard MIDI connectors are 5 way (180 degree) DIN plugs and sockets. Provided you use the appropriate type of socket connected in the manner outlined in Figure 2, the 'THRU' box can be wired into the system using standard MIDI leads. If you make up your own leads, twin screened cable is required. Pins 2, 4, and 5 on one plug are connected to the corresponding pins of the other plug, with the screen carrying the connection between the two pin 2s. Note that some audio 5-way DIN leads use cross coupling and are unsuitable for MIDI applications.

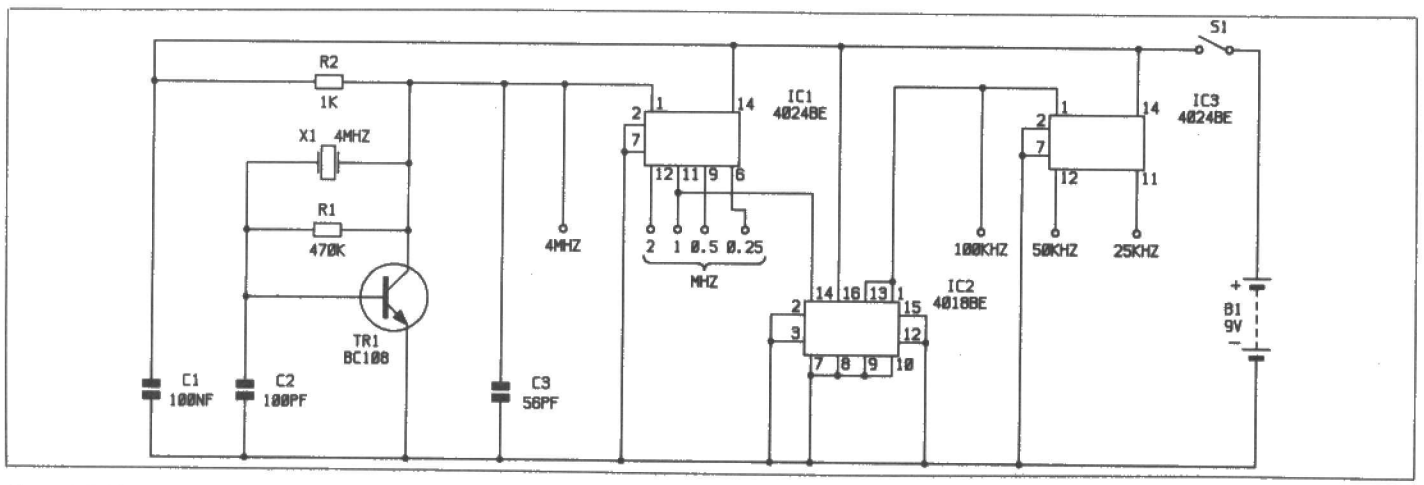
Crystal Calibrator

Some years ago a crystal calibrator offering a wide range of fundamental frequencies was a costly piece of equipment. Things have gradually changed, with the more popular crystals now costing what is in real terms probably less than one tenth of their cost ten or more years ago. Also, logic integrated circuits are now at virtually give away prices, and with digital dividers it is possible to generate a wide range of output frequencies from a single crystal oscillator.

This unit has a fundamental output at 4MHz, with additional outputs at 2MHz, 1MHz, 500kHz, 250kHz, 100kHz, 50kHz, and 25kHz. This may seem of limited value if the unit is to be used as a calibration oscillator for a shortwave radio, but bear in mind that these are only the fundamental signals, and that harmonics are available at frequencies to beyond 30MHz. Thus, for example, the 1MHz output also provides signals at 2MHz, 3MHz, 4MHz, 5MHz and so on. The outputs from 25kHz to 2MHz are all good quality squarewave types. In theory, the even harmonics (e.g. 2MHz, 4MHz, 6MHz with a 1MHz fundamental) are absent from a squarewave signal, but in practice these frequencies are generally relatively weak rather than absent, due to slight imperfections in the waveform. A pulse shaper could be added at the output in order to give a better spectrum of harmonics, but this is probably not worthwhile. The even harmonics will be of more than adequate strength without the aid of any pulse shaping. Because the output is a squarewave, the unit is also well suited to use as a timebase calibrator for an oscilloscope. It gives times ranging from 500ns per cycle at the 2MHz output to 40ms per cycle at the 25kHz output.



Crystal Calibrator.



Crystal Calibrator Circuit.

The circuit is quite conventional, and has TR1 as a crystal oscillator operating at 4MHz. A non-buffered 4MHz signal is available direct from the collector of TR1. C2 could be changed to a preset type to permit trimming of the output frequency. For most purposes this is unnecessary, as the frequency error is likely to be no more than a couple of Hertz if no trimming is used, and few applications genuinely require even this degree of accuracy. The 4MHz signal is fed to a CMOS 4024BE seven stage binary ripple counter (IC1). Only four stages are actually used in this circuit, and these produce outputs at 2MHz, 1MHz, 500kHz, and 250kHz. The 1MHz output is fed to the input of IC2. This is a CMOS 4018BE divide by 'N' counter, and in this circuit it is connected to give divide by ten operation. This gives the 100kHz output signal. This signal is also fed to the input of another 4024BE ripple counter (IC3). Only the first two stages of this are used here, and these provide the 50kHz and 25kHz output signals. Of course, the other outputs of IC3 could be used if desired, and they give output frequencies of 12.5kHz, 6.25kHz, etc. However, in practice these lower frequencies are likely to be of limited value. The eight outputs can be taken to separate sockets, or for greater convenience can be taken to a single socket via an eight way switch (which must be a break before make type).

Construction of the unit should not prove to be particularly difficult, but remember that all three integrated circuits are CMOS types, and that as such they require the standard anti-static handling precautions to be observed. Also, care should be taken when connecting X1, as overheating due to keeping the bit in place on the joints too long could impair the accuracy of the unit.

When coupling the output to a shortwave radio no direct connection should be made between the two units. Apart from possibly overheating the output of the unit, this would almost certainly give an excessive signal level at the receiver. An adequate signal transfer can usually be obtained by connecting a short lead to the output of the calibrator, and placing this close to another lead connected to the aerial input of the receiver.

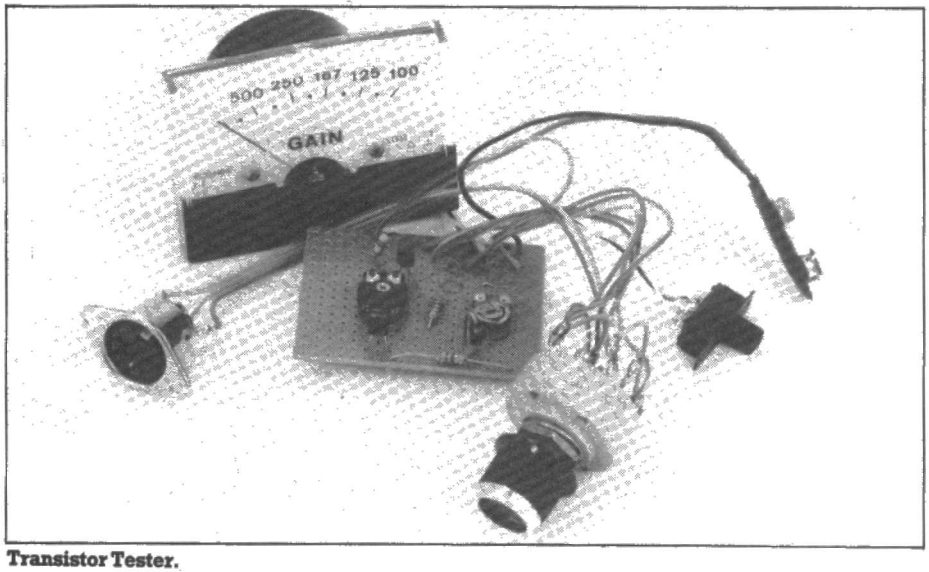
Transistor Tester

A conventional transistor tester operates by feeding a fixed base current to the test device and then measuring the collector current. The higher the gain of the device under test, the higher the current registered on the meter. The meter can therefore be easily calibrated in terms of current gain, and the collector current is in fact proportional to the gain of the test transistor. In practice this type of tester is very quick and easy to use, but is does have one slight flaw. This is that transistors are tested for gain at an unpredictable collector current, and the gain of most devices varies considerably with changes in collector current. What generally happens is that high gain devices produce a large collector current that means that are tested under good conditions, whereas low gain devices are tested at low currents where they perform relatively badly. This tends to exaggerate the differences between transistors, with slightly above average gain devices showing up as having very high gains. Perhaps of greater importance, transistors with fairly low gains can tend to show up as having inadequate current gains, where they are in fact up to specification in this respect.

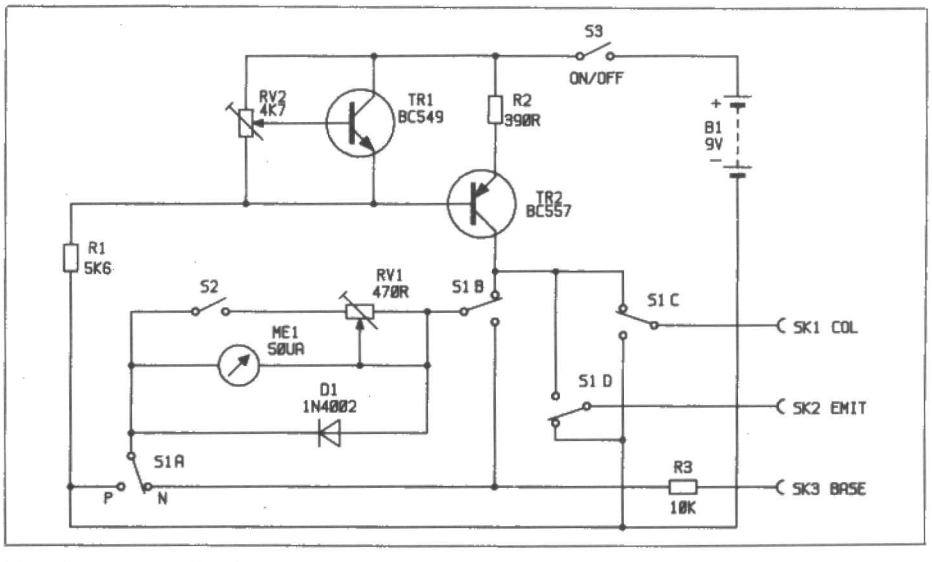
This transistor tester was designed to be as easy to use as the conventional tester of the type outlined above, but to operate at a constant collector current so that more reliable results are obtained. Results with the device have been encouraging, and it seems to give what are more accurate results than those obtained with other ultra-simple testers.

The basic method of operation is to feed the collector and emitter terminals from a constant current source. The base terminal is fed from the same source but via a sensitive current meter. With this arrangement the collector current is always equal to the figure set by the current source (or something very close to this figure if we allow for the small amount of current tapped off through the base circuit). The base current is the amount needed to produce this collector current, and therefore decreases as the level of current gain increases. The meter reading can therefore be translated into a measure of current gain. A minor drawback of this arrangement is that it gives a reverse reading and non-linear scale, which is less convenient in use than the forward reading linear type of a conventional tester. However, taking readings from this tester is no more difficult than taking readings from the similar scale of an analogue multimeter's resistance ranges.

There is little more to the circuit than the constant current generator based on TR1 and TR2. RV2 is adjusted to give an output current of 5 milliamps, which is a



Transistor Tester.



Transistor Tester Circuit.

good compromise value that suits most types of transistor. S1 switches the meter and the test sockets to suit both npn and pnp devices. The meter has a full scale value of 50 microamps, which gives a gain range of one hundred at full scale deflection to infinity at zero deflection. Closing S2 increases the full scale value of the meter to 500 microamps, and the gain range is then ten at full scale deflection, again running through to infinity at zero deflection. R3 and D1 protect the meter against severe overloads, such as would otherwise occur if the base and emitter leads of the test device are connected but the collector is not.

Construction is fairly straightforward, but take care to avoid errors in the switching. Probably the best type of switch to use for S1 is a 3 way 4 pole rotary type, with the adjustable end-stops set for 2 way operation. SK1 to SK3 can be 1 millimetre sockets grouped close together, and most transistors will then plug into these without any difficulty. Some types (especially power types) will not, and a set of test leads fitted with crocodile clips will be needed to connect these into circuit.

In order to adjust RV2 for a 5 milliamp output current, connect a multimeter switched to a d.c. current range (about 10 to 25 milliamps full scale) between the negative supply rail and the collector of TR2. Then adjust RV2 for the correct reading of 5 milliamps. To give RV1 the correct setting, close S2 and then connect the current meter across the base and emitter test sockets via a 15k resistor. Switch the multimeter to a more sensitive range (about 1 to 2.5 milliamps full scale), and then adjust RV1 to match the two readings (bearing in mind that ME1 now reads 500 microamps full scale).

It is more than a little helpful to give the meter a new scale, and this is not too difficult using rub-on transfers. Meter movements are quite delicate though, and due care needs to be taken if you decide to do this. The current gain for points on the scale is obtained by dividing 5000 by the scale value (200 at a value of 25uA for example). When the unit it switched to the lower range, gain levels will be one tenth of the indicated value.

CAPACITANCE METER ADAPTOR PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,3	100k	2	(M100K)
R2	1M	1	(M1M)
R4	10k	1	(M10K)
R5	1k	1	(M1K)
R6	6k8	1	(M6K8)
R7	150k	1	(M150K)
RV1	47k Hor Sub-min Preset	1	(WR60Q)
RV2	1k Lin Pot	1	(FW00A)
CAPACITORS			
C1	100μF 10V PC Electrolytic	1	(FF10L)
C2	47nF Polyester	1	(WW37S)
C3	4μ7F 63V PC Electrolytic	1	(FF03D)
C4,5	100nF Ceramic	2	(YR75S)
SEMICONDUCTORS			
IC1,2	TLC555CP	2	(RA76H)
IC3	μA78L05	1	(QL26D)
MISCELLANEOUS			
SK1	1mm Socket Red	1	(WL60Q)
SK2	1mm Socket Black	1	(WL59P)
SK3	2mm Socket Red	1	(HF47B)
SK4	2mm Socket Black	1	(HF44X)
S1	4-way 3-pole Rotary Switch	1	(FF75S)
S2	SPST Ultra-min Toggle Switch	1	(FH97F)
B1	9V PP3 Battery	1	(FK62S)
	Battery Connector	1	(HF28F)
	8-pin DIL Socket	2	(BL17T)

TRANSISTOR TESTER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	5k6	1	(M5K6)
R2	390Ω	1	(M390R)
R3	10k	1	(M10K)
RV1	470Ω Sub-min Hor Preset	1	(WR54J)
RV2	4k7 Sub-min Hor Preset	1	(WR57M)
SEMICONDUCTORS			
TR1	BC549	1	(QQ16R)
TR2	BC557	1	(QQ16S)
D1	1N4002	1	(QL74R)
MISCELLANEOUS			
S1	3 way 4 pole Rotary Switch	1	(FF76H)
S2,3	SPST Ultra-min Toggle	2	(FH97F)
ME1	50uA Panel Meter	1	(FW98G)
B1	9 volt PP3 Battery	1	(FK62S)
SK1,2,3	1mm Socket	3	(WL59P)
	Battery Connector	1	(HF28F)

MIDI THRU BOX PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1,5,6,9,10,13, 14,17,18	220Ω	9	(M220R)
R2,7,11,15	2k2	4	(M2K2)
R3,8,12,16	4k7	4	(M4K7)
R4	2k7	1	(M2K7)
CAPACITOR			
C1	10μF 25V Axial Electrolytic	1	(FB22Y)
SEMICONDUCTORS			
IC1	6N139	1	(RA59P)
TR1,2,3,4	BC559	4	(QQ18U)
MISCELLANEOUS			
SK1,2,3,4,5	5-way (180°) DIN Socket	5	(HH34M)
S1	SPST Ultra-min Toggle	1	(FH97F)
B1	Battery 1.5V	4	(FK55K)
	8-pin DIL Socket	1	(BL17T)
	Battery Holder	1	(HF29G)
	Battery Connector	1	(HF28F)

CRYSTAL CALIBRATOR PARTS LIST

RESISTORS: All 0.6W 1% Metal Film			
R1	470k	1	(M470K)
R2	1k	1	(M1K)
CAPACITORS			
C1	100nF Ceramic	1	(YR75S)
C2	100pF Ceramic	1	(WX56L)
C3	56pF Ceramic	1	(WX53H)
SEMICONDUCTORS			
IC1,3	4024BE	2	(QX13P)
IC2	4018BE	1	(QX10L)
TR1	BC108C	1	(QB32K)
MISCELLANEOUS			
B1	9 volt PP3 Battery	1	(FK58N)
S1	SPST Ultra-min Toggle	1	(FH97F)
X1	4MHz HC-18/U	1	(FY82D)
	Battery Connector	1	(HF28F)
	DIL IC Holder 14 pin	2	(BL18U)
	DIL IC Holder 16 pin	1	(BL19V)

Exploring Radio

by Graham Dixey C. Eng., M.I.E.R.E. Part 4

Introduction

This radio makes use of the very versatile ZN414 radio chip but, unlike the little personal receiver, whose construction was described in the last issue of 'Electronics', this one has a rather more 'gutsy' output. This is obtained by the use of an audio amplifier chip, the TBA820M. This has a power capability of up to about 1.2 watts of audio power, depending upon the supply voltage and the choice of speaker. It is quite happy with any supply voltage between 3V and a maximum value dependent upon the speaker impedance, as follows.

4 ohm speaker – 9V maximum
8 ohm speaker – 12V maximum
16 ohm speaker – 16V maximum

However, in this design, the d.c. supply has been set at 9V so that any speaker with a coil impedance of 4 ohms or greater can be used. The prototype, shown in Photo 1, used an 8 ohm speaker and gave excellent results. The recommended value of supply voltage should be adhered to because it is related also to the design of the d.c. feed to the ZN414.

A further feature of this receiver is that it is dual waveband, with a choice of either medium-wave or long-wave listening selected with a slide switch. Photo 2 shows the fully assembled pcb.

The Ferrite Rod Aerial

Two separate coils are wound on a 100mm long ferrite rod. The medium wave coil consists of 55 turns of 30 swg enamelled copper wire (E.C.W.). These should be 'close wound', i.e. no gaps between turns. This coil should be wound fairly close to one end of the rod so as to leave plenty of room for the long wave coil. The latter consists of 250 turns which are 'scramble wound'. This means

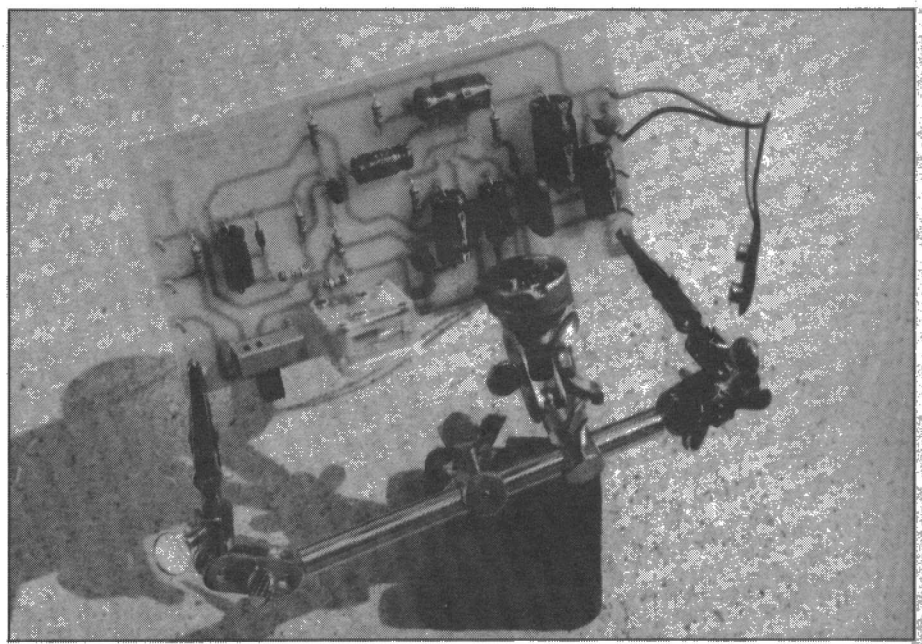


Photo 1. The prototype TRF receiver under construction. A jig of this type greatly helps in handling PCBs.

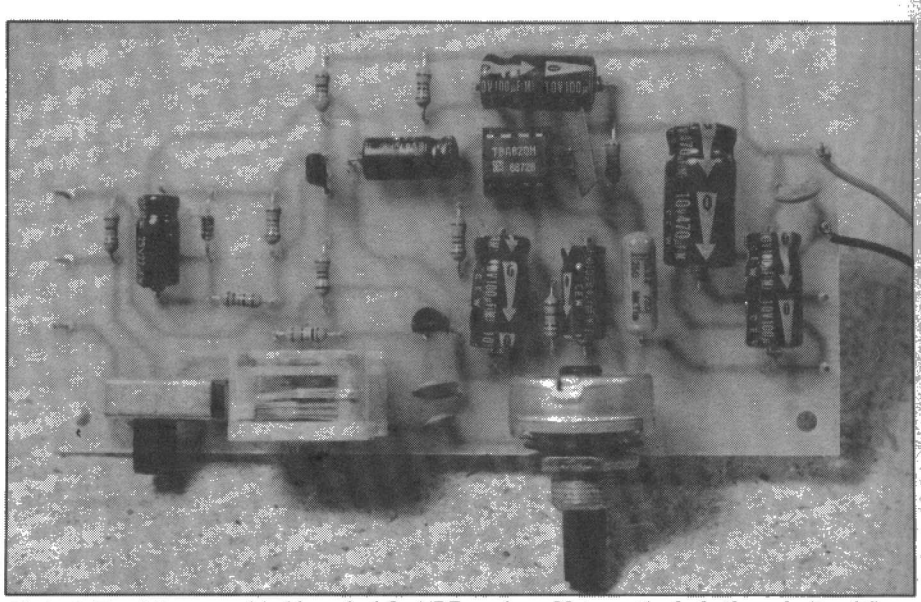


Photo 2. The fully assembled board of the TRF receiver. Note particularly the polarity of the electrolytic capacitors and the placement of other components.

that all attempts at neatness are cast aside (they're impossible anyway without a coil winding machine) and the turns are simply wound tightly over each other. If two or three centimetres of space are allowed for this coil and the turns, as they are laid on, are wound on first in one direction and then in the other, the general result should be a fairly uniform coil with just a hump in the middle (see photograph 3). The coils can be secured with short lengths of PVC tape but, in the example shown, 'superglue' was used instead. The usual precautions should be observed when using this adhesive, of course.

The Radio-Frequency Section

Looking at Figure 1 you can see that each of the coils is selected as required by a slide switch. A 220k resistor is connected across the long wave coil to improve the frequency response. Tuning is by a 150pF variable capacitor, which is provided by using the 'aerial' section of an AM Varitune tuning capacitor assembly. The input stage is decoupled by the usual filter consisting of a 100k resistor and a 10nF ceramic disc capacitor (R2 and C1). A 100nF ceramic disc decouples the output of the ZN414. This output connection also serves as the d.c. feed point for the ZN414 and is provided by a drive circuit, based around the ZTX300 transistor TR1. The supply voltage is dropped from 9V to 4.7V by R6, this value being stabilised by the zener diode ZD1. The emitter resistor, R5, of TR1 also sets the level of the Automatic Gain Control (A.G.C.) for the ZN414.

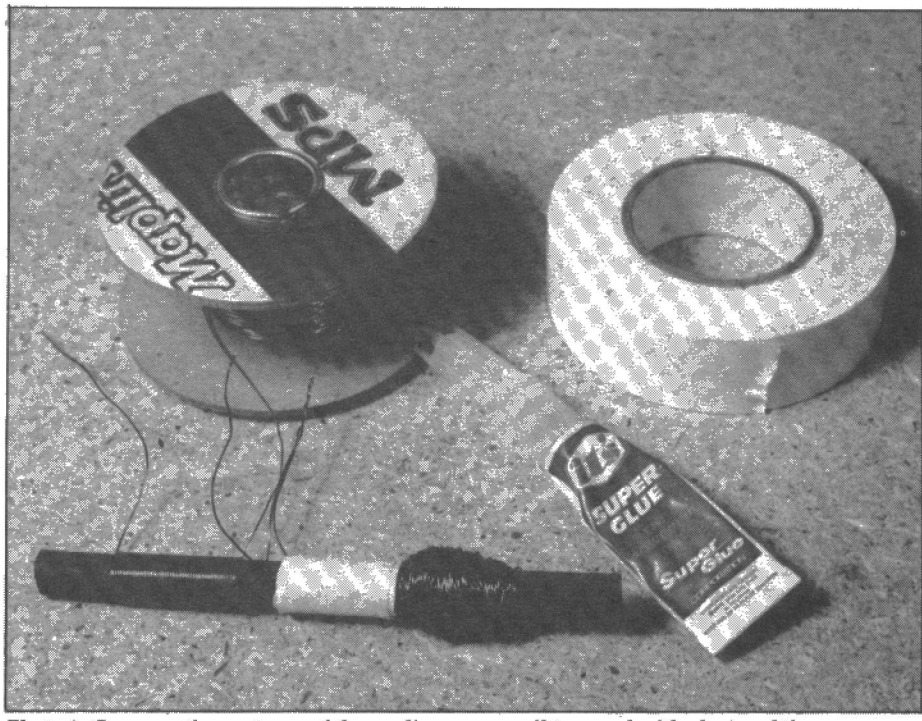


Photo 3. Compare the neatness of the medium wave coil (secured with glue) and the 'scramble wound' long wave coil (secured with a piece of PVC tape).

The Audio Amplifier

The output of the ZN414 is developed across the volume control RV1 and is coupled to the input of IC2 by C4. This chip has a number of discrete components associated with it, which establish its operating conditions and ensure its stability. The arrangement shown was found to be completely stable and gave an output of acceptable quality, bearing in mind the limitations of the fairly restricted bandwidth of a.m. transmissions anyway. The audio output is coupled to the speaker through a 470μF capacitor, C10. The d.c. supply to the receiver is decoupled by the capacitors C11 and C12.

General Comments

In the case of the capacitors C11 and C12, some readers may wonder why a small value capacitor (C12 = 100nF) should be included in parallel with a very large value capacitor, in this instance C12, which is 100μF. This is a situation which occurs in many circuits where high frequencies or high speeds (i.e. digital circuits) are used. The low value capacitor will invariably be specified as a 'ceramic disc' type while the larger capacitor will be found to be of a totally

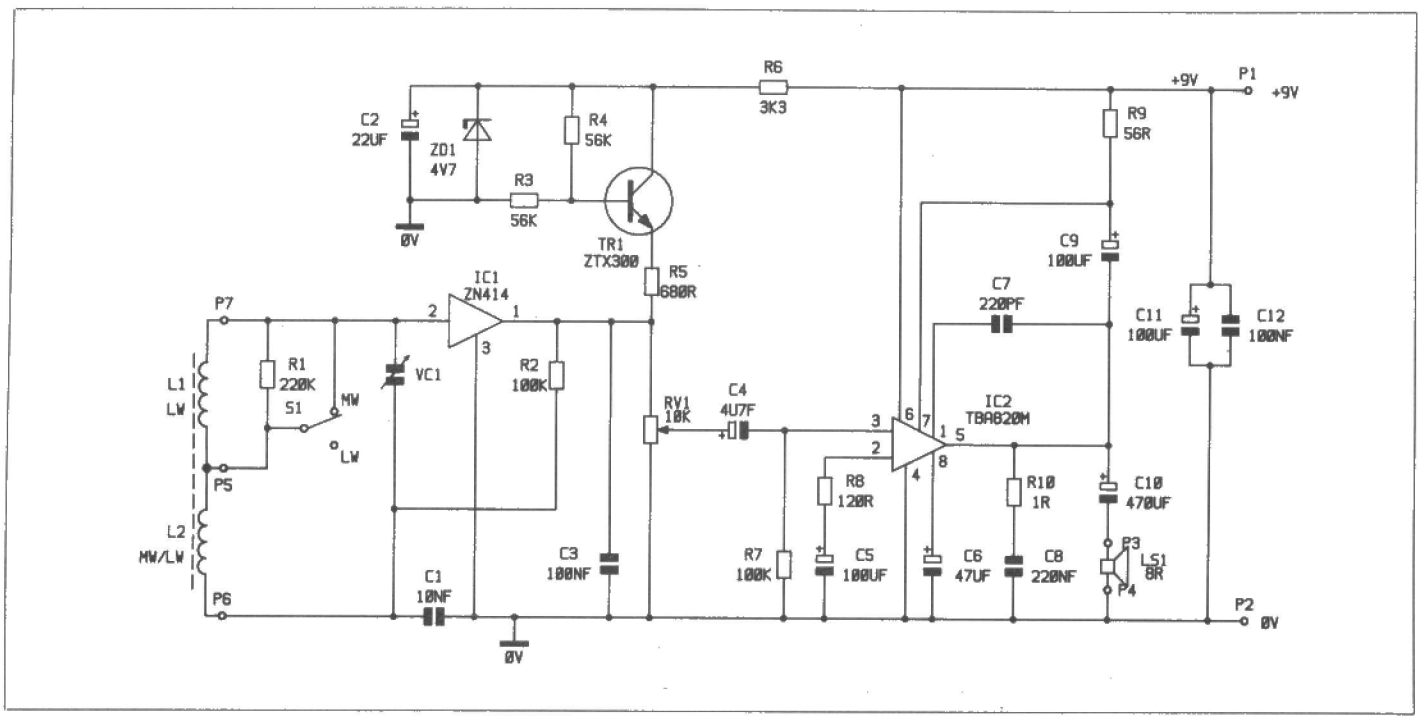


Figure 1. Circuit of the MW/LW TRF Receiver.

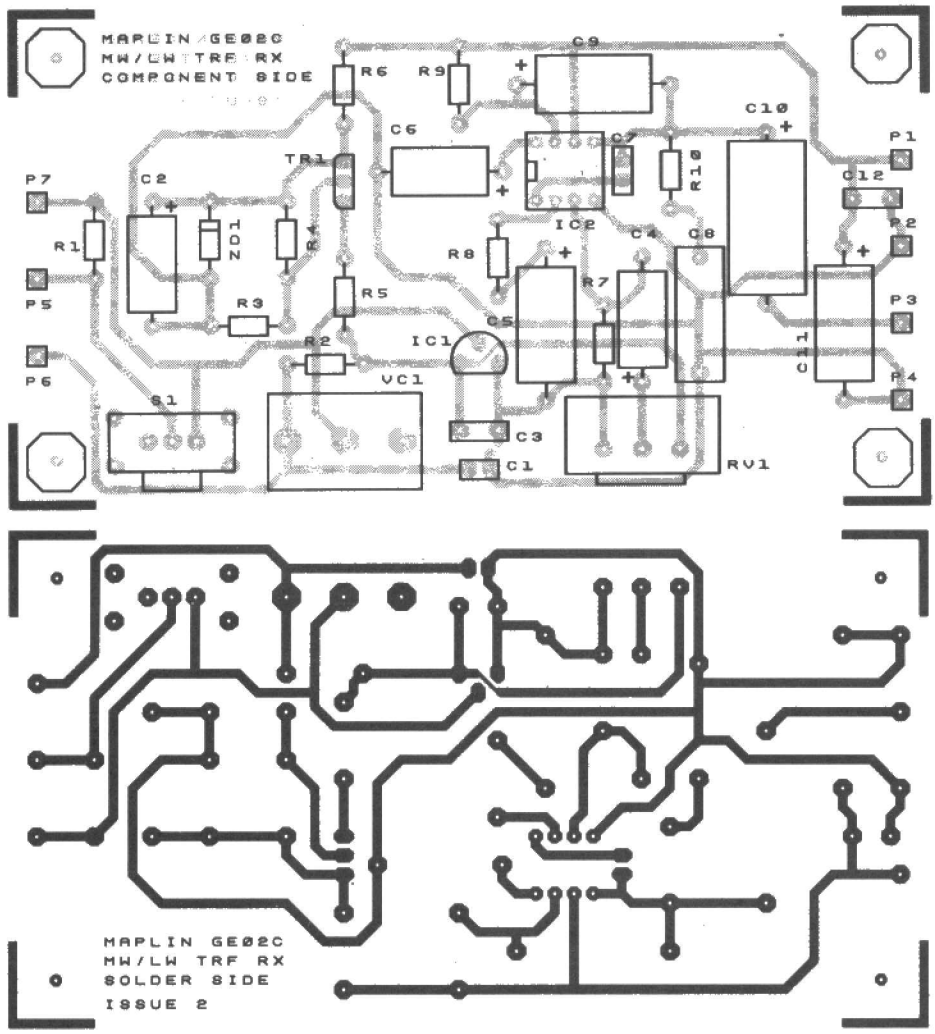


Figure 2. Layout of the PCB.

different type, as in this case where it is an electrolytic. At high frequencies capacitors should be as 'pure' as possible, that is they should exhibit only the properties of capacitance and nothing else. However, because of the way in which electrolytics are made (and this also applies to polyester capacitors), they also have a certain amount of 'self-inductance'. While this would have little effect at audio frequencies, its presence at radio frequencies would prevent the capacitor from acting effectively as a decoupling or r.f. 'bypass' capacitor, simply because it would have too much reactance to the radio signal. For this reason, whenever a ceramic disc capacitor is specified in a circuit, it should never be replaced by a polyester type, just because the value is the same in both cases. One 100nF capacitor is not necessarily the same as another 100nF capacitor!

Assembling and Testing the Receiver

The availability of a PCB for this project, see Figure 2, will ensure both ease of construction and instant success when the power is applied for

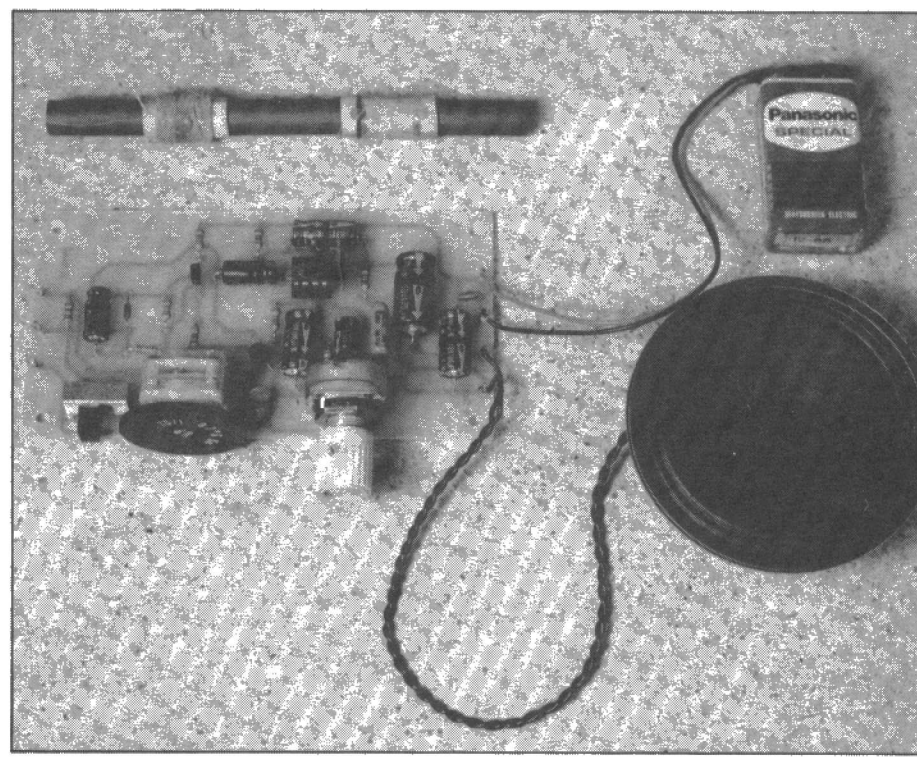


Photo 4. The completed TRF receiver, shown here with the fully assembled 'alternative' ferrite rod aerial.

the first time. It is merely assumed that the builder will check the following.

(a) the components are inserted in the correct places in the board.
(b) the polarity of the zener diode and the electrolytic capacitors is observed.
(c) the base connections of the ZN414 and ZTX300 are correctly noted and these components wired in correctly; check and double check before applying power.
(d) the TBA820M is plugged into its base the correct way round.
(e) all soldered joints are made cleanly without any dry joints or 'bridges' between adjacent tracks.

Having observed all of the foregoing points, the power can be applied and the result should be a working radio, Photo 4 shows the completed unit. The only controls are the waveband switch, the tuning capacitor and the volume control, all of which will obviously now be checked.

Should the situation be less than happy, the voltage checks shown in Table 1 can be made. These were made on the prototype, under 'no signal' conditions with a 20k/volt multimeter on the 10V d.c. range. Visual checks should also be made on the components in case one is incorrectly placed.

The readings obtained in practice may differ slightly from those in Table 1 but large variations from these values will indicate a fault condition.

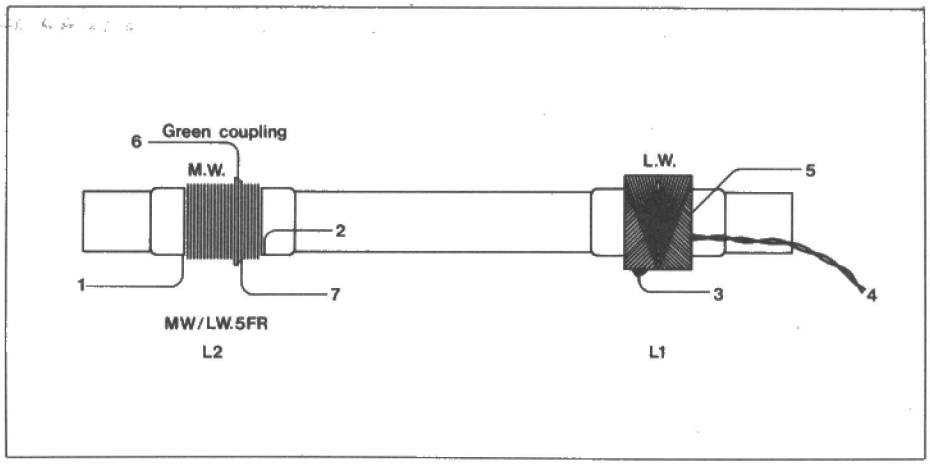


Figure 3. Ferrite rod aerial connections.

Test Point	Voltmeter Reading
Collector of TR1	4.0V
Base of TR1	1.65V
Emitter of TR1	1.2V
Output pin of ZN414	0.85V
Input pin of ZN414	0.6V
Pin 1 of TBA820M	0.6V
Pin 2	0.5V
Pin 3	0V
Pin 4	0V
Pin 5	4.15V
Pin 6	9V
Pin 7	8.8V
Pin 8	5.6V

Table 1.

An Alternative Ferrite Rod Aerial

The ferrite rod aerial specified at the start of this article gives an adequate performance on medium waves but not such a good performance on long waves. This situation can be improved by using a proprietary aerial assembly, such as the one available from Maplin Electronics, and it is listed as an alternative component in the Parts List. It can be used merely by disregarding both the small green coupling coil and the tap on the long wave coil. A drawing of this aerial assembly is included, see Figure 3, and it will be seen that all that needs to be done is to connect the two coils in series by strapping connecting wires 2 to 3 (and taking this junction to pin P5 on the board); wire 1 then goes to pin P6 and wire 5 goes to pin P7. This assembly has the further advantage that it is possible to move the coils along the rod to find the best position; they can then be fixed with a blob of glue.

MW/LW TRF RECEIVER PARTS LIST

RESISTORS: All 0.6W 1% Metal Film

R1	220k	1	(M220K)
R2,7	100k	2	(M100K)
R3,4	56k	2	(M56K)
R5	680Ω	1	(M680R)
R6	3k3	1	(M3K3)
R8	120Ω	1	(M120R)
R9	56Ω	1	(M56R)
R10	1Ω	1	(M1R)
RV1	10k Lin Pot	1	(FW02C)

CAPACITORS

C1	10nF Ceramic Disc	1	(YR73Q)
C2	22μF 25V Electrolytic	1	(FB30H)
C3,12	100nF Ceramic Disc	2	(YR75S)
C4	4μ7F 100V Electrolytic	1	(FB18U)
C5,9,11	100μF 10V Electrolytic	3	(FB48C)
C6	47μF 16V Electrolytic	1	(FB38R)
C7	220pF Ceramic	1	(WX60Q)
C8	220nF Polyester	1	(BX76K)
C10	470μF 10V Electrolytic	1	(FB71N)
VC1	AM Varitune	1	(FT78K)

SEMICONDUCTORS

TR1	ZTX300	1	(QL46A)
ZD1	BZY88C4V7	1	(QH06G)
IC1	ZN414	1	(QL41U)
IC2	TBA820M	1	(WQ63T)

MISCELLANEOUS

L1,2	Ferrite Rod 810	1	(YG20W)
LS1	8Ω Speaker	1	(YW53H)
S1	Slide Switch R/A	1	(FV01B)
	PP3 Battery Clip	1	(HF28F)
	EC Wire 30 swg	1	(BL40T)
	Pin 2145	1 Pkt	(FL24B)
	PCB	1	(GE02C)
	Hook-up Wire	1 Pkt	(BL00A)
	8-pin DIL Socket	1	(BL17T)

ALTERNATIVE

L1,2	Ferrite Rod Aerial	1	(LB12N)

The following item is available, but is not shown in our 1988 catalogue:
MW/LW TRF Receiver PCB, Order As GE02C
Price £3.95

CALCS

by Jeff Scott Part 1

Complex Numbers

Were you bored by maths at school? Unfortunately, the teaching of maths, not only in schools but also in engineering colleges, tends to be a pure science instead of showing the practical applications to electronics and other branches of engineering.

Mathematics is an exact science with the answers popping out like a pudding out of a pudding bowl, as my teacher used to put it. The frontiers of mathematics, the philosophical side usually predicts the trend that engineering is likely to take.

For instance, the large scale digital processing of signals was shown mathematically long before the technology was available. Fourier analysis showed the components in a waveform much before spectrum analysers were available. Explosions and massive gravitational forces in space, compressing material into a black hole were explained mathematically before the physical phenomena were understood.

Mathematics also serves the purpose of manipulating data into the required form. For example, Laplace transforms transfer an expression from the frequency domain into the time domain. This is useful in electronics since old methods of analogue amplification and filtering were in the frequency domain (frequency division multiplex) whereas modern techniques are in the time domain (time division multiplex).

Complex numbers is an important branch of mathematics with applications to electronic engineering. Mathematicians use the letter 'i' in complex numbers, but 'j' is used in electronics since the letter 'i' in electronics is reserved for current. The 'j' symbol is often called the 'j' operator. The need for a 'j' operator arises for the following reasons. In a circuit that is purely resistive the current is in phase with the voltage, see Figure 1. In a circuit that is reactive, the current may lead or lag the voltage depending on whether the circuit is inductive or capacitive, see Figures 2 and 3.

Argand, a mathematician, was the first to invent the use of 'i' or the imaginary operator. This was useful for explaining the square roots of negative numbers. Consider the equation $x^2 - 16 = 0$. The solution is $x = \pm 4$ which is represented on an Argand diagram as in Figure 4.

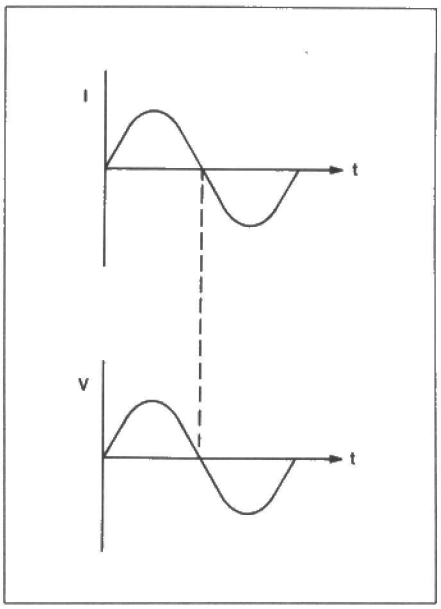


Figure 1. Current in phase with voltage.

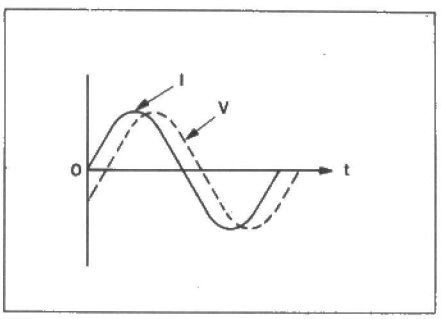


Figure 2. Current leading voltage.

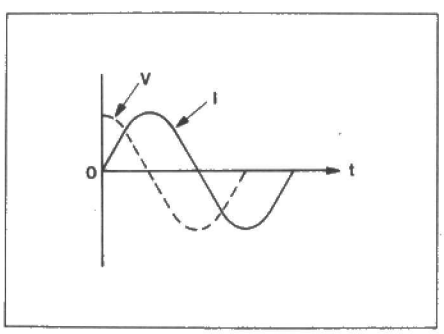


Figure 3. Current lagging voltage.

But how does one solve the equation $x^2 + 16 = 0$?

$$\text{Here } x^2 = -16$$

$$x = \pm 4\sqrt{-1}$$

There is no real number which represents the square root of -1. Argand's solution to this problem was to assign real values along the x axis and imaginary values along the y axis. The solution of $x^2 = -16$ is now shown in Figure 5.

In electrical engineering there are no imaginary values but we call these quadrature

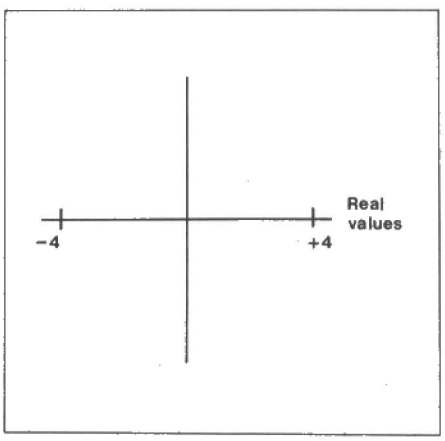


Figure 4. Argand diagram.

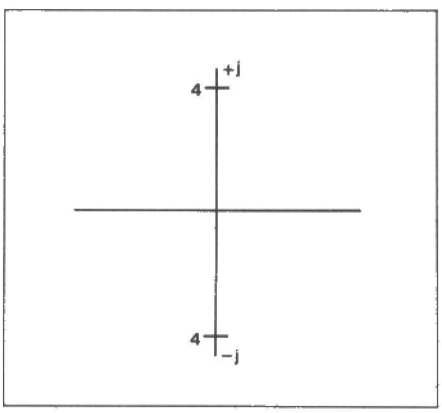


Figure 5. Imaginary (quadrature) components.

components. Therefore a purely resistive circuit will have real components only and a reactive circuit will have quadrature components. An Argand diagram has facilities for describing a circuit that has both real and quadrature components.

From now on we shall use the j symbol and examine the addition and subtraction of complex numbers. An expression that has both real and quadrature components is called a complex number. For instance a number that has 2 units of real value and 4 units of quadrature value would be represented as in Figure 6. The vector P is represented in the general form of: $a + jb$, or $2 + j4$ in this case.

The above method of representing a complex number uses the Cartesian co-ordinates. There is another method of representing complex numbers using polar co-ordinates. In this method, if we define an angle θ (Figure 7) from a reference point then we have one of the attributes defined.

The other attribute must be the length of the vector. Both quantities can be calculated from the cartesian co-ordinates. From Pythagoras' theorem the hypoteneuse in Figure 7 is given by:

$$\sqrt{a^2 + b^2}$$

or $$\sqrt{20^2 + 15^2}$$

$$= \sqrt{400 + 225}$$

$$= \sqrt{625}$$

$$= 25$$

The tangent of angle θ is $\frac{15}{20} = 0.75$

Therefore $\theta = 36.87°$

The resultant is sometimes called the modulus and the angle is called the argument.

Addition and Subtraction

One of the advantages of the complex notation is that numbers are added and subtracted quickly and easily. This is done by adding or subtracting the real components separately from the quadrature components. The alternative is the tedious method of drawing parallelograms and finding the resultant of only two vectors at a time.

Figure 8 (not to scale) shows three complex numbers $2 + j2$, $-5 + j8$, $-10 - j4$. Adding the real components gives -13 and adding the quadrature components produces $j6$. Therefore the answer is $-13 + j6$. Subtraction is just as easy compared to the parallelogram method of operating on only two vectors each time.

Multiplication and Division of Complex Numbers

Before we deal with the multiplication of complex numbers we must see what happens when we multiply j by itself successively. Since:

$$j = \sqrt{-1}$$

$$j^2 = -1$$

$$j^3 = -1 \times j = -j$$

$$j^4 = -1 \times -1 = +1$$

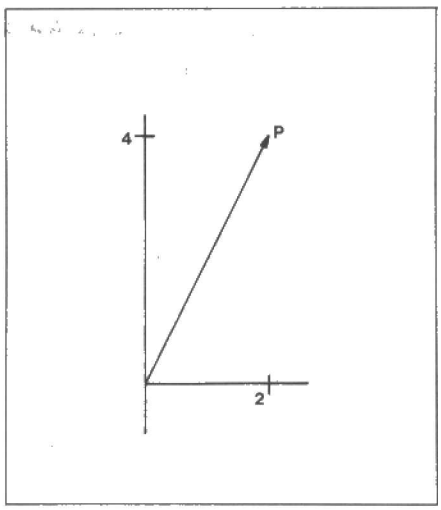


Figure 6. Cartesian co-ordinates.

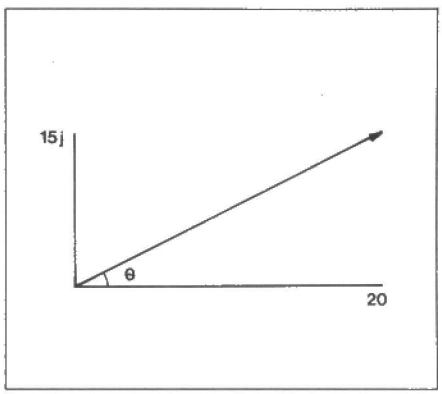


Figure 7. Polar co-ordinates.

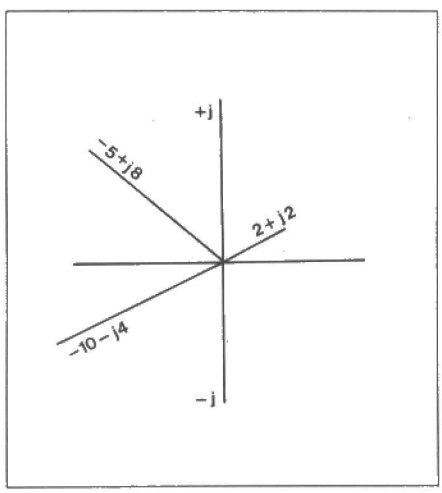


Figure 8. Three complex numbers.

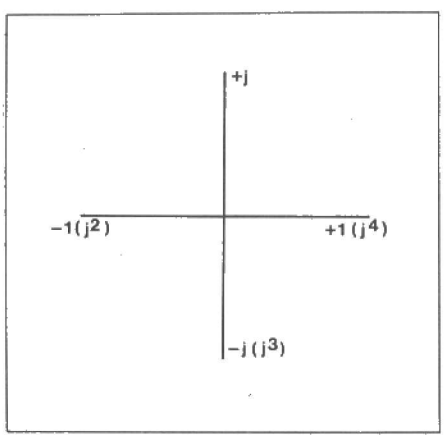


Figure 9. The 'j' operator.

Now we can see why it is called an operator. From Figure 9, each successful multiplication by j has the effect of swinging the value through 90 degrees.

In order to multiply two complex numbers together, we simply expand the brackets in the usual manner.

For example: $(3 + j^5)(4 - j^2)$

$$12 + j20 - j^2 2$$

Remembering $j^2 = -1$,

$$12 + j20 + 2$$

$$14 + j20 \text{ is the result.}$$

Division of complex numbers is just as simple, the answer dropping out like a pudding out of a pudding bowl.

For instance $\frac{3 + j5}{4 - j2}$

First we rationalise the denominator by multiplying it by its conjugate. The conjugate has the same values but opposite phase and has the effect of turning into a wholly real number. The conjugate of $4 - j2$ is $4 + j2$.

So $$\frac{(3 + j5)(4 + j2)}{(4 - j2)(4 + j2)}$$

$$= \frac{12 + j6 + j20 + j^2 10}{16 + j8 - j8 - j^2 4}$$

$$= \frac{12 + j26 - 10}{16 + 14}$$

$$= \frac{2 + j26}{20}$$

$$= 0.1 + j1.3$$

Multiplication and Division using Polar Co-ordinates

Polar co-ordinates lend themselves quite easily to multiplication and division. The modulus is multiplied separately and the angles (argument) are added together in multiplication:

$r_1 \angle\theta_1 \times r_2 \angle\theta_2 = r_1 \times r_2 \angle\theta_1 + \theta_2$

$6 \angle 35 \times 7 \angle 40 = 42 \angle 75$

For division the moduli are divided and the angles are subtracted:

Continued on Page 46.

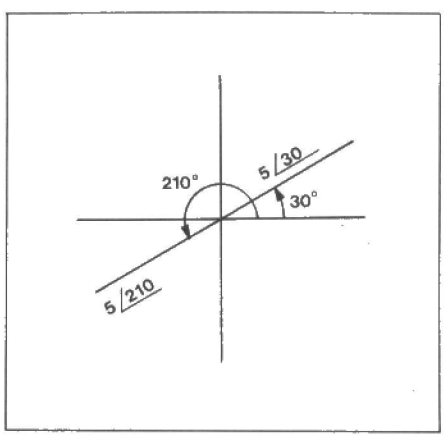


Figure 10. Roots of a number.

NOT FLAT JUST RUN DOWN

by Stephen Potts

Over the past few years the use of Nickel Cadmium or Nicad batteries has become increasingly popular. However, many people are not getting full use from their batteries. This article should help you to understand what the Nicad cell can do when treated properly. It also shows you how to test whether your cell is in a healthy state and if not what can be done about it.

The Nicad cell is a secondary cell because, like a car battery, it can be recharged. The sort of cell that cannot be recharged is known as a primary cell.

Nicads are made in two forms; sintered plate and pocket plate. These are then subdivided into those that are vented and those that are not vented.

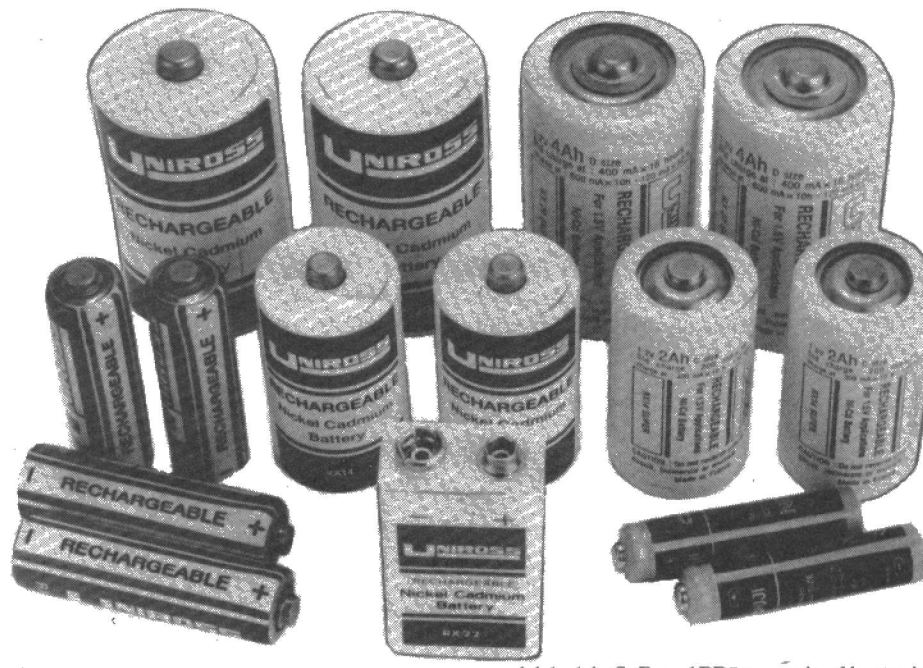
Types of Cells

The sintered plate design has the greatest power density. That is, more power can be placed in a small volume and they should last for up to 500 cycles, or 10 years, whichever is the greater. However, they frequently have much shorter lives because of maltreatment and life times of a mere 100 cycles are not uncommon.

The sintered cell Nicad has, like all Nicads, a voltage of 1.25 volts and a capacity that depends on the size of the cell. The capacity is measured in amp-hours and is a product of the time in hours that the cell can deliver a certain current, measured in amps. Thus a cell that is rated at 1.2 amp hours could be expected to deliver 0.6 amps for 2 hours. The typical cell capacities are 0.5 amp-hours for AA cells, 1 to 2.2 amp-hours for C cells and 1.2 to 4 amp-hours for D cells.

Whereas the sintered cell Nicad has active ingredients spread uniformly about the cell, the pocket cell Nicad has the active ingredients in a fine web of pockets. Since these pockets prevent the active material migrating, the cell tends to have a longer life than the sintered glass variety. Coupled with this is a lower energy density and a higher internal resistance. The pocket cell variety will recharge in a shorter time. If instructions supplied with a battery say; 'recharge for less than three hours', then it is probably a pocket plate cell. If the recharging time is less than 1 hour, then it is definitely a pocket plate cell.

Pocket plate cells are available in C and D sizes, but not usually any smaller. They have a life of 500 to 1000 cycles or 15 years if they are sealed, and a life of 1500 to 2000 cycles or 20 years if vented.



AAA, AA, C, D and PP3 type nicad batteries.

Electrical Characteristics

Having looked at the types of cell available we can now look at the characteristics of the cell. As stated before, the cell has an output voltage of 1.25 volts which is significantly lower than that of the lead acid cell at 2.2 volts. This may not appear significant, but if you wish to make up a 12 volt pack, it is 6 lead acid cells, but 10 Nicad cells. It is for this reason that many people underestimate the number of cells they are going to need.

Unlike most other cells, the Nicad does not give much warning that it is running down. The voltage will remain approximately 1.25 volts for the full life of the charge and drop very rapidly as it runs down. The computer program accompanying this article will show this. The point at which voltage drop occurs is usually when around 10% to 20% of the cell capacity is left. Any further discharge – if this

happens on a regular basis – will harm it, shortening its life and lowering its efficiency.

There is a case where it is of use to drain the cell completely, and this is when the cell has 'memory'. Cells that are frequently shallow cycled (that is; fully charged, used for a very short time before being charged again) develop lazy habits and will not deliver more than a shallow capacity. An example of this is the wall mounted torch, often used by the front door of a house. If it is discharged every night as the occupier goes out for about 10 minutes to put the cat out, the result of doing this over a period of time, is that the torch will now deliver no more than 10 minutes of light. A way round this is to exercise the cell. This is achieved by fully discharging the cell at a rate of C/2 or more, where C is the capacity of the cell. Thus a 0.5 amp-hour cell will be discharged at 0.25 amps. The cell is then rapidly recharged, with care being taken not to overcharge it. This is repeated 5 times, and in many cases it will rejuvenate tired cells.

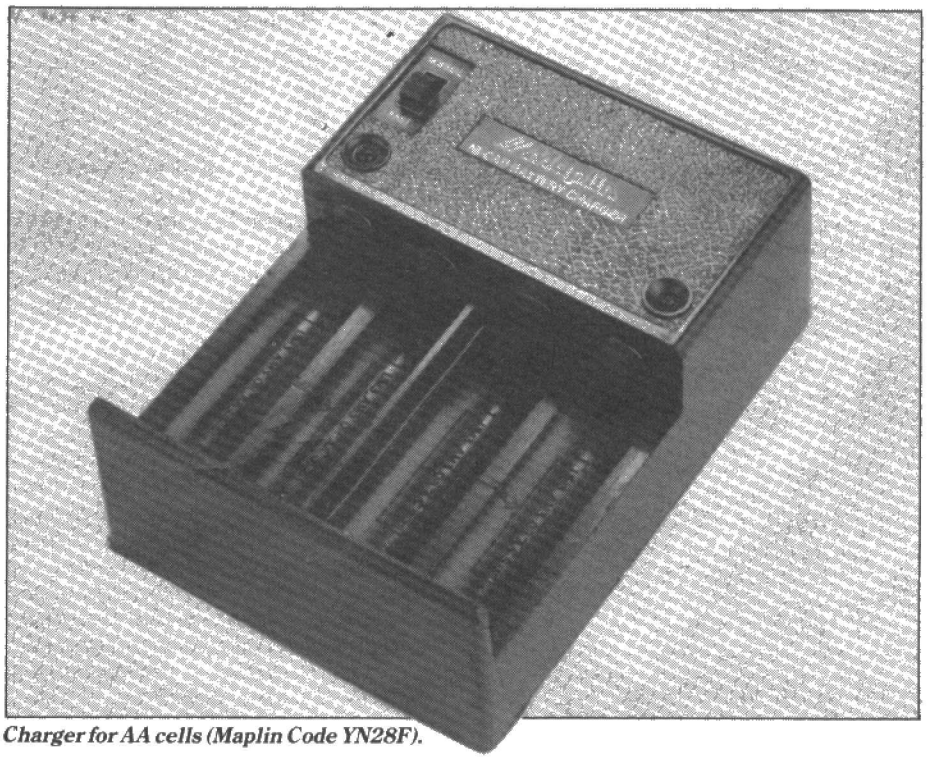
There is another cause of tired cells and that is dendritic growth, which is the growth of fine crystals in the cell. These short-out in the cell, and the cell will appear as a dead short and have an output voltage, when charged, of less than 0.8 volts, and less than 1 volt when being charged. The cell can be rejuvenated, but it requires a current of around 200 amps for a fraction of a second with consequent risk of the cell exploding and taking eyes and skin with it. Personally, I would rather throw out the cell and start again.

Correct Charging

The cells can cause injury problems in another way. They can discharge their full capacity in less than a minute, and a cell in your pocket, with some keys and small change, can short circuit and cause burns.

The most frequent cause of cells being damaged is the recharging process. To get the maximum out of the cells they should be recharged as soon as possible. They should either be recharged at a high rate and not overcharged, or alternatively recharged at between C/4 and C/20. If recharging is done in a very short time, then care should be taken that the cell is not overcharged since this will lead to overheating and permanent damage to the cell because of its low internal resistance. The ideal charging rate is C/4 to C/20. If C/20 is used, then the cell will tolerate overcharging at room temperature since the excess heat is easily lost. The important thing is that the cell is charged at a *constant current and not constant voltage.* Figure 1 shows a possible charging circuit. If the cells to be charged are 2 AA cells, then they could be charged at 0.05 amps for 12 to 14 hours. Since they will have a voltage of 2.5 volts, the net voltage across the cells will be 12 minus 2.5 volts or 9.5 volts. A resistor of 190Ω will provide this.

Maplin Electronics have several different models of Nicad battery chargers in their catalogue ranging from a simple AA charger to a sophisticated 'deluxe' charger which accepts AAA, AA, C, D, PP3 and Button cells. It is highly recommended that charging of batteries is carried out using this type of proprietary commercial product as all safety precautions have been built into the units.



Charger for AA cells (Maplin Code YN28F).

Testing Cells

How do you know whether your cells are in a reasonable state or not? This was the problem I had with several AA cells. There were no markings to show which cells were the old ones and which were the new ones. The following computer routine was developed to overcome this problem.

The BBC computer has an analogue to

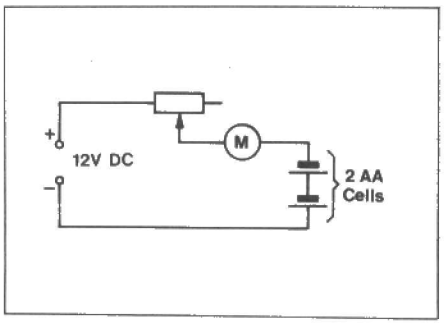
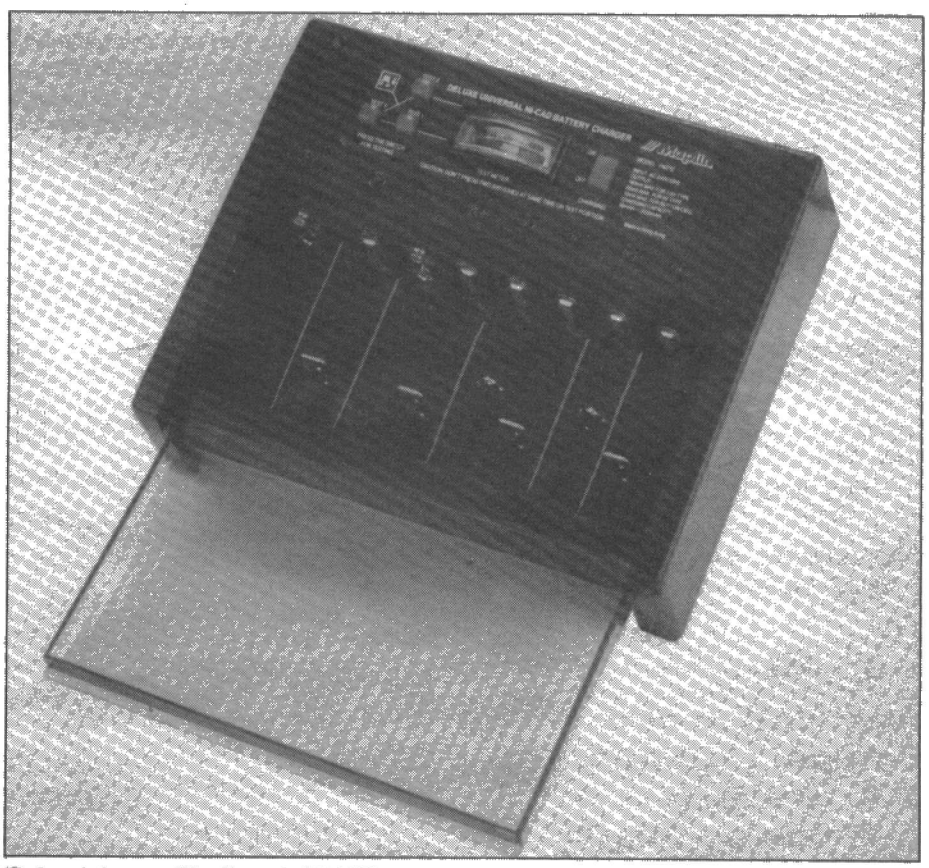


Figure 1. Simple charging circuit.



'Deluxe' charger (Maplin code YN27E).

digital converter (the joystick port) which will accept a voltage of up to 1.8 volts and produce a signal in proportion to the voltage. This makes life very easy. The signal from the port will be approximately 35168 for 1 volt input. The advantage of using the computer is that it can store the results and a graph of the results after several hours can be produced. The principle behind the circuit in Figure 2 is that the cell is discharged through a low value resistor and the computer is used to monitor the voltage.

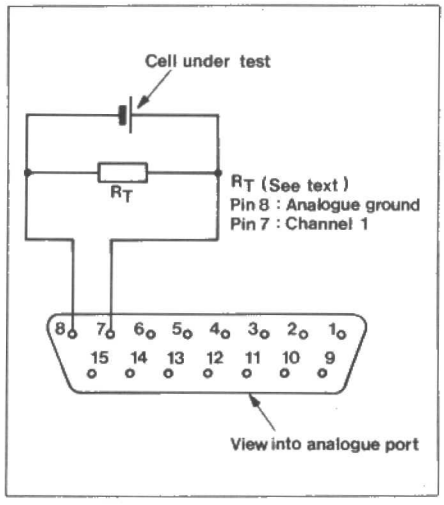


Figure 2. Circuit for BBC micro.

The value of the resistor that is used to discharge the cell should be calculated to produce a current of C/2, C/4 and C/8 to give a spread of figures. Thus for an AA cell of capacity 0.5 amp-hours, the discharge rate at C/2 is 0.25 amps, the resistor will have to be 1.25/0.25 or 5Ω. For C/4 the resistor will be 10Ω and so on. In each case, it is a good idea to place an ammeter in the circuit to check that the cell is discharging at the correct rate. In practice, it is often a problem to obtain resistors of these values and it is quite satisfactory to set up a circuit with several resistors wired in parallel to obtain approximately the correct current, as long as the same block of resistors is used when comparisons are made several months later.

The program to check the performance of the cell is shown in Table 1, it is very simple and could be much enhanced. Possible improvements include an automatic graphing routine and a method of storing the information on each cell so that a comparison can be made at a later stage in the cell's life.

Line 150 stops further readings being taken when the voltage drops below approximately 1 volt. It does not stop the cell being further discharged, though by using relays switched from the user port, it could turn off the cell when the voltage drops too low.

```
10  REM Ni Cad cell monitor.
20    DIM Rdg%(1000)
30    REM Readings stored in Rdg%()
40    PRINT "Enter the time interval for
      readings in minutes."
50    INPUT Min
60    Gap=Min*60*100
70    Co=0
80    REPEAT
90          TIME=0
100         Co=Co+1
110         REPEAT
120         UNTIL TIME>Gap
130         Rdg%(Co)=ADVAL(2)
140         PRINT Co*Min,
      INT(Rdg%(Co)/35.168)/1000
150   UNTIL Rdg%(Co)<35000
160   REM This stops the routine when
      the voltage falls too low
170   FOR C=1 TO Co
180   PRINT C*Min,
      INT(Rdg%(Co)/35.168)/1000
190   NEXT C
```

Table 1. Monitor program.

I hope this monitor program will be useful to you and that you have learnt something from this article about the correct way of caring for your Nicads.

Calcs continued from Page 43.

$$r_1 \angle\theta_1 \div r_2 \angle\theta_2 = \frac{r_1}{r_2} \angle\theta_1 - \theta_2$$

$25 \angle 60 \div 5 \angle 40 = 5 \angle 20$

Polar co-ordinates are particularly useful in finding square roots. The square root of the modulus is found and the angle divided in half:

$(r \angle\theta)\,½ = \sqrt{r} \angle\theta/2$

$(25 \angle 60\,½ = 5 \angle 30$

Now a real number like 25 has two roots 5 and −5.

These are of the same magnitude but opposite sign (phase) to each other. Similarly, a complex number has two roots, one 180 degrees out of phase (opposite phase) to each other.

Hence $(25 \angle 60)½$ has roots of $5 \angle 30$ and $5 \angle 210$, see Figure 10. To square a number in polar co-ordinates we square the modulus and double the argument.

$(r \angle\theta)^2 = r^2 \angle\theta \times 2$

$(4 \angle 20)^2 = 16 \angle 40$

Application of Complex Numbers to A.C. Bridges

We shall now see how many of the above methods are applied to the solution of equations for AC bridges. In an equation containing both real and imaginary (quadrature) terms, the real components can be equated separately from the quadrature components.

For instance:

$R + jwL = 13 + j19$

Therefore $R = 13$

and $jwL = j19$

Let us examine a more complicated case like the Maxwell bridge of Figure 11.

At balance $Z_1 Z_x = Z_2 Z_3$

where $Z_1 = R_1 \dfrac{(1)}{(jwC_1)} = \dfrac{R_1}{jwC_1R_1 + 1}$

$Z_2 = R_2$

$Z_3 = R_3$

$Z_x = R_x + jwL_x$

Substituting in $Z_1 Z_x = Z_2 Z_3$:

$$\frac{(R_1)}{(jwC_1R_1 + 1)}(R_x + jwL_x) = R_2R_3$$

Multiplying both sides of the equation by $(jwC_1R_1 + 1)$:

$R_1(R_x + jwL_x) = R_2R_3(jwC_1R_1 + 1)$

$R_1R_x + jwL_x = jwC_1R_1R_2R_3 + R_2R_3$

Equating real terms:

$R_1R_x = R_2R_3$

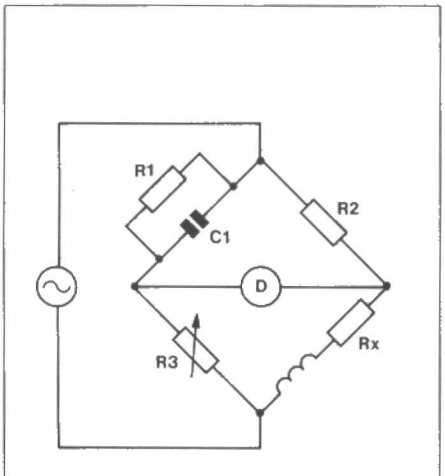


Figure 11. Maxwell bridge.

$$R_x = \frac{R_2R_3}{R_1}$$

Equating imaginary terms:

$wL_x = wC_1R_1R_2R_3$

$L_x = C_1R_1R_2R_3$

This enables one to calculate the value of an unknown inductor Lx and its associated resistance Rx from known values of the other components in the bridge. It is hoped that the above shows how even large mathematical problems can be tackled in small easy steps.

ELECTRONICS BY EXPERIMENT

Part 4 by Graham Dixey C. Eng., M.I.E.R.E.

Introduction

A register is also a sequential logic circuit but, whereas a counter is concerned with the process of 'counting' the input data, the register is principally concerned with storing it. However, this bold statement can be a little misleading, since it implies an event that is going on for a substantial period of time, which may not be the case at all. As an example of this consider the familiar electronic calculator. When two numbers are added together, each is deposited into a register and on the command 'add' the two numbers are replaced by their sum. This sum is held in a register that only a moment before held one of the numbers. Thus, the storage of the original number was of very short duration. Another example is when data is being sent from a microcomputer to a printer. It passes out via a register, each byte in turn being held by this register until it is replaced by the next one in the queue. Thus, a register is really a string of flip-flops that can hold a data word for a period of time, no matter how long or short that is. There are also certain other ways in which particular types of register can be useful and they will also be discussed.

To start with the most elementary idea of a register, consider Figure 1, in which the four possible modes of operation are shown. These modes arise because there are two ways in which data can be transmitted from point to point. These ways are known as 'parallel' and 'serial' transmission respectively.

In parallel transmission, there is one line for each bit of the data word and all bits are transmitted at once. Thus an eight-bit word (a byte) requires eight separate lines – an example of parallel data transmission is the well-known Centronics printer interface. Because all bits are sent at once, parallel transmission is fast but requires lots of conductors.

In serial transmission, there is one line only, which all bits have to share. They are therefore sent down it one at a time. Thus, no matter how many bits there are in the data word only one transmission line is used but the process gets slower and slower as the bits queue up for their turn on the line! The advantage is the simplicity of it. An example of serial transmission is the RS232 interface for printers and modems.

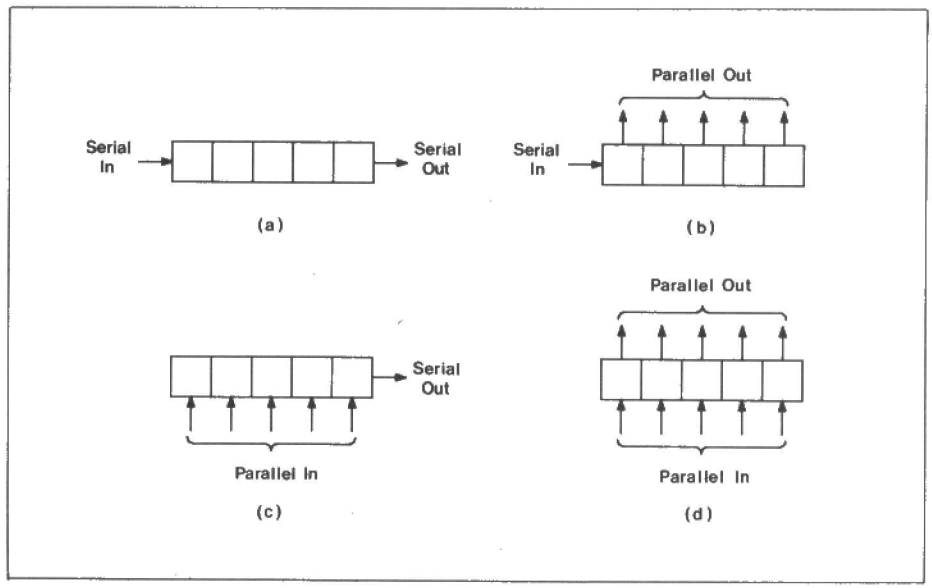


Figure 1. The four main modes of register operation. (a) Serial In, Serial Out (SISO); (b) Serial In, Parallel Out (SIPO); (c) Parallel In, Serial Out (PISO); (d) Parallel In, Parallel Out (PIPO).

How does this relate to registers? Quite simply. Each flip-flop of a register holds just one bit of the data word. Serial or parallel operation refers to the ways in which the data is loaded into the register or sent out from it.

For example, one mode is Serial In Serial Out (SISO). There is only one input line and only one output line. Data bits are 'clocked in' one at a time and then 'clocked out' in a similar fashion. For example, if the register is 'eight bits wide' (meaning it has eight flip-flops), it will take eight clock pulses to store a data word and a further eight clock pulses to 'shift' it out again. Note the use of the word 'shift'. Registers of this type are known as shift registers. In continuous operation a stream of bits flows in at one end and out at the other.

However, it is possible to shift the data in serially until all flip-flops hold one bit of the data word and then access all bits of the data word at once, so that the output is 'parallel'. Obviously this mode of operation is known as Serial In Parallel Out (SIPO).

Reversing the latter mode is also possible. With one clock pulse the whole register can be loaded with the data word, which can then be shifted out serially. This is known as Parallel In Serial Out (PISO) operation.

A little thought will reveal that the SIPO type is a 'serial to parallel' converter, while the PISO type is a 'parallel to serial' converter. The significance of this can be seen where a computer is connected either to another computer or to a peripheral through a serial connection, i.e. a single line. Since the internal organisation of a computer is on parallel lines (referring to the data bus), there must be a parallel to serial conversion at the sending end and a serial to parallel conversion at the receiving end. It is an ideal function for registers to perform.

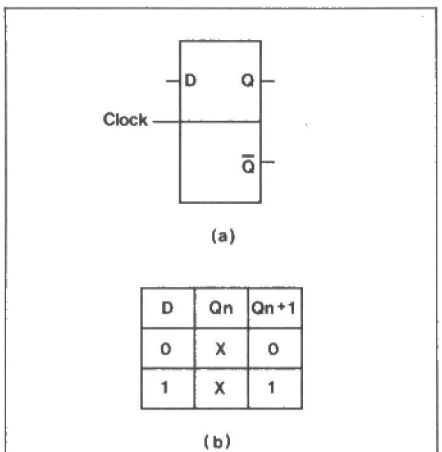
Finally, it is possible to load a register by parallel input and access it also by parallel mode. This is Parallel In Parallel Out (PIPO).

The D-type Flip-flop

As a general (but not invariable) rule, counters use JK flip-flops and registers use a type known as the D flip-flop (D stands for Data). The D flip-flop is rather simpler in operation then its JK equivalent. Both its symbol and truth table are shown in Figure 2. It has a single input line, marked D, a CLOCK input and a pair of complementary outputs, of which the Q output is the more useful. The truth table reveals its true simplicity.

The logic level, logic 0 or logic 1, to be stored is first applied to the D input. When a clock pulse is applied this level at D shifts to the Q output, where it will remain until a new data bit is applied to D and a further clock pulse arrives. Thus, the Q output merely copies whatever is at the D input after the arrival of a clock pulse.

As an experiment, try connecting a low-frequency clock oscillator to the clock input of a D flip-flop, say 1Hz (a suitable design appeared in Part Three). At the same time connect a switched logic level to the D input. Connect a logic level indicator to the Q output. Randomly vary the logic level input and note how, on the arrival of the next clock pulse, the logic level applied to D transfers to Q.



D	Qn	Qn+1
0	X	0
1	X	1

Figure 2. The D type flip-flop. (a) Circuit symbol; (b) Truth Table showing that, after clocking, the output at Q equals the input at D. Note: Qn = state of Q before clock pulse; Qn+1 = state of Q after clock pulse; X = 'Don't care', i.e. can be 0 or 1.

The 7474 D-type Flip-flop

This package contains two D flip-flops and its pin-out diagram appears in Figure 3. This is an example of a 'positive edge triggered flip-flop'. This was mentioned in Part Three but will be repeated here. The clock pulse can be considered as two transitions occurring one after the other. The first is from Logic 0 to Logic 1 (the positive edge) and the second is from Logic 1 back to Logic 0 (the negative edge). In a positive edge triggered flip-flop only the first transition matters. The data transfer from D to Q occurs at this instant only. The duration of the clock pulse is of no importance, nor is the subsequent trailing edge.

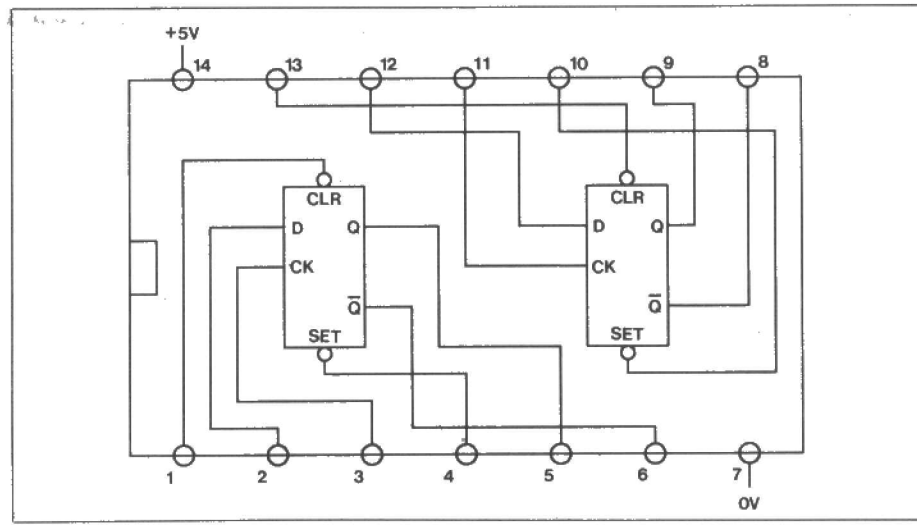


Figure 3. Pinout diagram of the 7474 Dual D type, positive edge triggered, flip-flop.

Each flip-flop has two connections not so far mentioned. These are marked on the 'chips' as SET and CLR and are known usually as 'preset' and 'preclear'. The 'bubble' on each of these indicates that they are 'active low' connections, that is taking the terminal to Logic 0 causes the required action to take place. It is usual to connect all the CLR pins together for all the flip-flops in a register and call this the RESET line. Taking this low, momentarily, clears all the flip-flops, that is all Q outputs go immediately to Logic 0. The SET pins can be used when required to preset a given value into a register, that is to give it a particular state to start off with rather than starting with a clear register. Examples of this use appear later.

Shift Registers

The simplest types of register are the SISO and SIPO types. Their circuit diagrams appear in Figure 4. Basically they are the same; it is only in the way they are used that they differ. The data input to the register is the D input of the first flip-flop; the data input of any subsequent flip-flop is always the Q output of the preceding one. All flip-flops are clocked at the same time.

Purely for simplicity, only four-bit registers are shown. The same principles apply however many stages there are. To study the mode of operation, connect up this register using two 7474 ICs (Figure 5). Connect all CLR pins together and wire this common line through a switch to the 0V line; leave the switch open for the moment. Connect a de-bounced switch (see Part Three for details) to the clock input and logic level indicators to each of the Q outputs. Finally connect a logic level switch to the data input. Switch on; the Q outputs will be quite arbitrary. Momentarily connect the RESET line to 0V using the switch provided for this purpose. The Q outputs should now all be zero. Set the logic level at the input to Logic 1. No changes should be observed in any of the flip-flops. Leave this level as set and pulse the de-bounced switch four times. The Q outputs should take up the successive

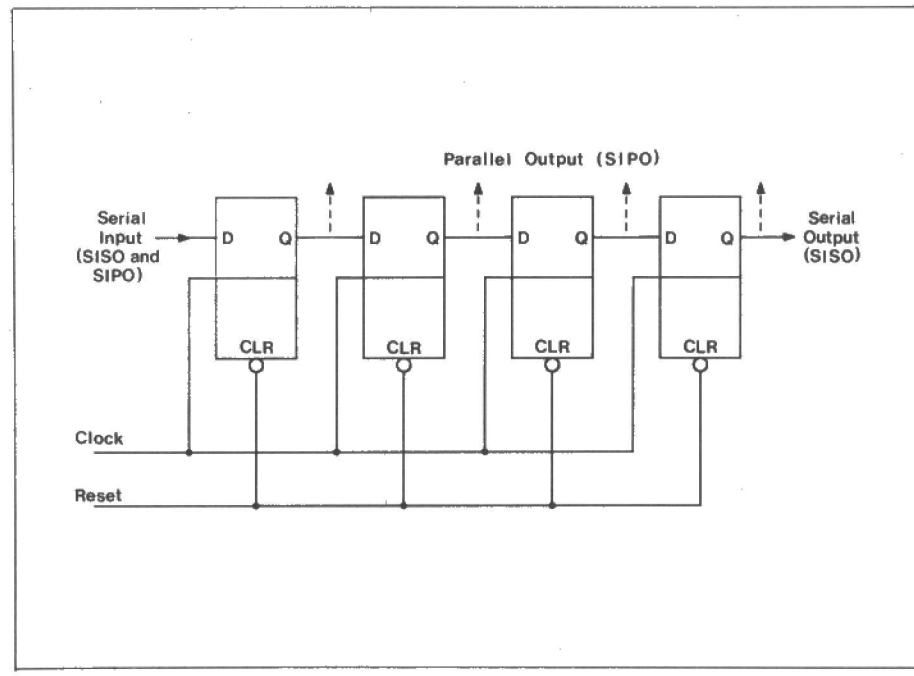


Figure 4. A 4 bit Serial In, Serial Out shift register (SISO), with the simple modification needed to get a parallel output (SIPO).

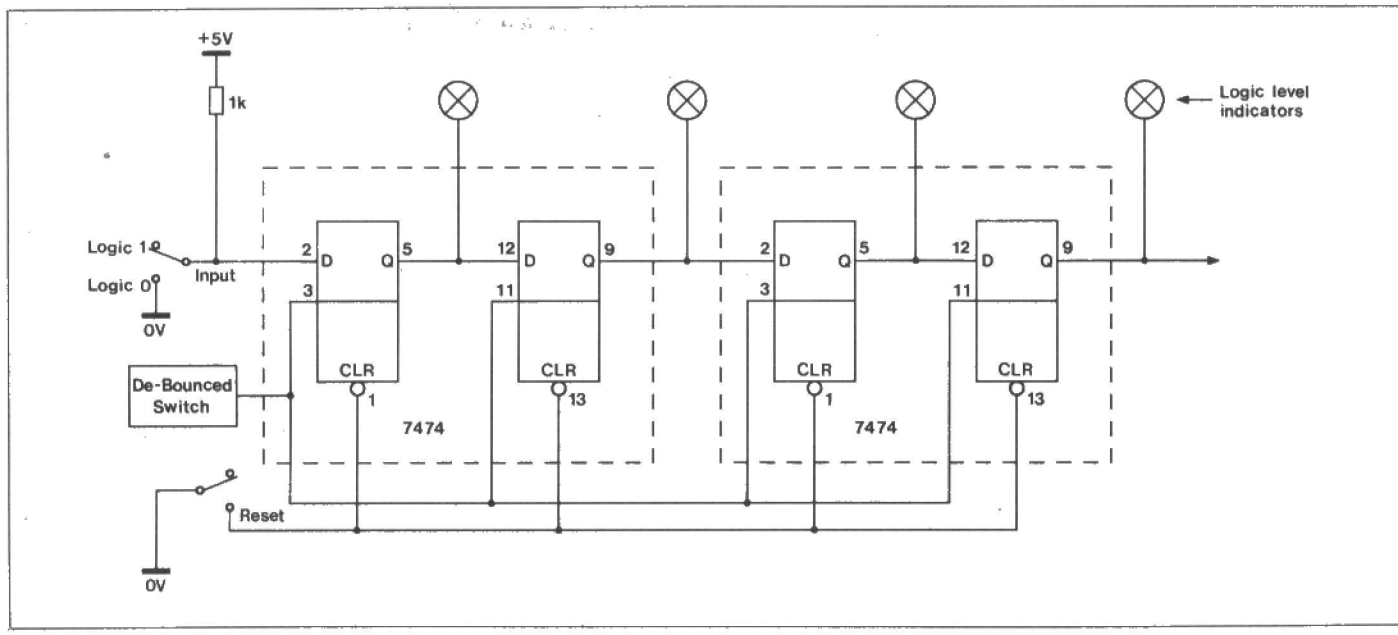


Figure 5. The circuit of Figure 4 implemented with two 7474 ICs and the necessary hardware for evaluating it.

states: 1000, 1100, 1110 and 1111, as shown by the four LEDs of the logic level indicators. If it doesn't do this, check in turn – the circuit wiring, the ICs themselves, the output from the de-bounced switch; finally, "have you left the RESET switch low?"

If all is well, consider the fact that the data word 1111 is shown by the four LED indicators. In other words, not only has it been stored but it is available at the Q outputs if four lines were to be connected to these outputs. Thus, although the data word was entered serially using four clock pulses, it is now available at the Q outputs without any further action being necessary. This is SIPO operation.

But if it is desired to remove the data word serially, four more clock pulses will be needed. This will normally load a new word so set the logic level input switch to Logic 0 and pulse the de-bounced switch four more times. The logic levels at the Q outputs should now take up the successive states: 0111, 0011, 0001 and 0000. Note that the original data word has been progressively shifted to the right until it was replaced by the new word 0000. This is an example of SISO operation. What has not been done in this case is to 'catch' the data word as it left the register. Special provision would have to be done to do this; nonetheless, the basic operation of serial shifting has been shown.

Parallel In Parallel Out and Parallel In Serial Out Registers

These registers, shown in Figure 6, illustrate that sometimes extra gating is needed to get the register to perform in particular ways. In these cases, each flip-flop requires a pair of two-input NAND gates and an inverter. These are concerned with the parallel loading. Once the parallel load has taken place, the contents of the register can be accessed in parallel mode merely by connecting wires to all of the Q outputs, or the data can be obtained serially by clocking the register four times, as already described for the

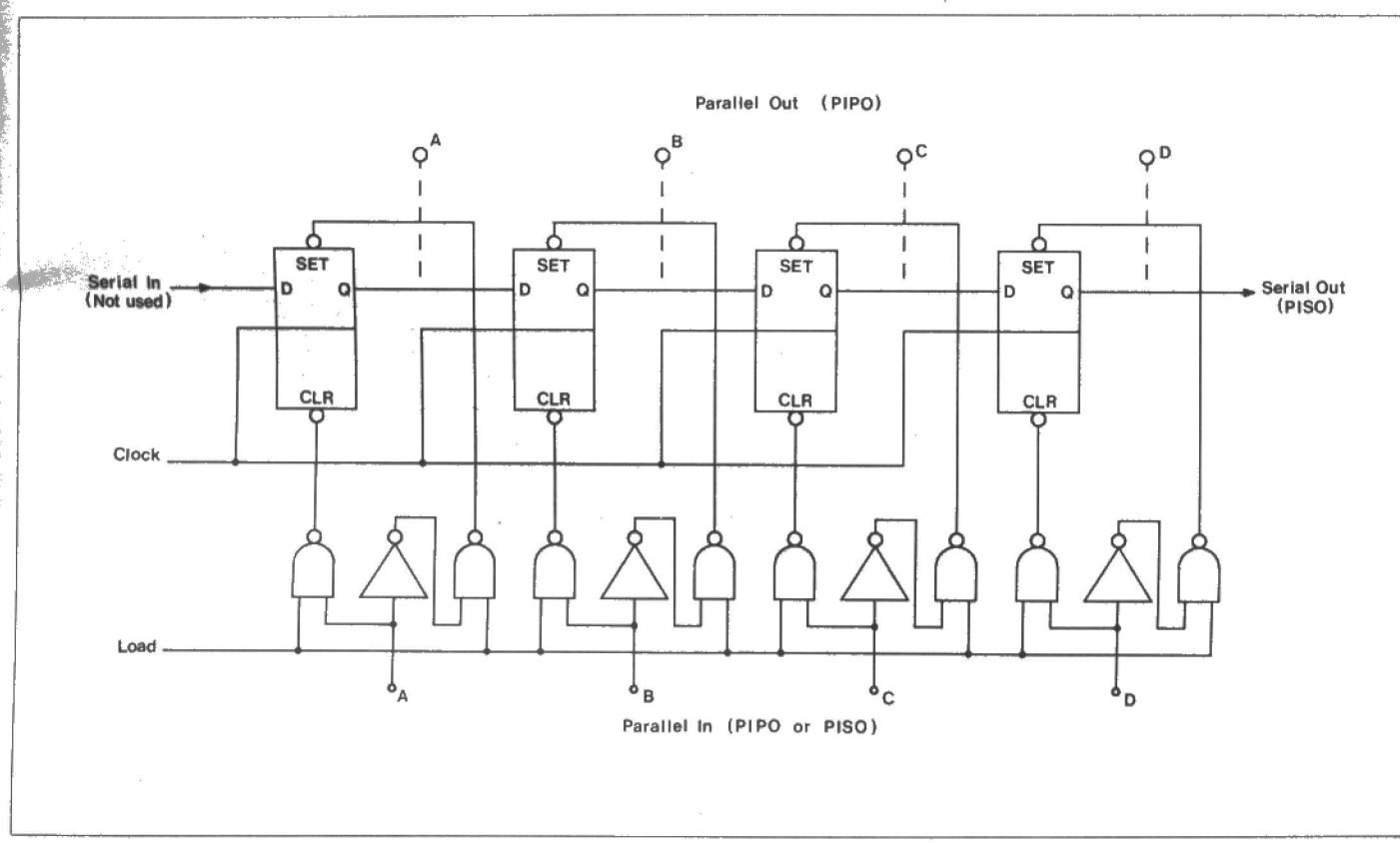


Figure 6. Circuit diagram(s) for the Parallel In, Serial Out (PISO) and the Parallel In, Parallel Out (PIPO) registers.

SISO and SIPO types.

The parallel loading works as follows. The line marked LOAD is normally held at Logic 0. Thus, both NAND gates of each pair associated with each register stage have logic 1 output levels. These drive the SET and CLR pins of the flip-flops; both pins being at logic 1s, nothing happens. Each DATA IN line will have either a Logic 0 or a Logic 1 on it, according to the data to be loaded. This is applied directly to one NAND gate and in inverted form to the other NAND gate. Consequently, one NAND gate of a pair will have Logic 1 and Logic 0 at its inputs, while the other will have both inputs at Logic 0. Now let the LOAD line go to Logic 1. One of each pair of NAND gates will now have two Logic 1 inputs, the condition that causes the output of a NAND gate to go to Logic 0. Since the SET and CLR inputs are negative-acting, each flip-flop will become SET or CLEAR according to the logic level at its DATA IN input. The best way to follow this operation is to connect up the circuit and use a logic probe to look at the logic levels on all pins of the NAND gates and inverters during the load operation.

To connect up the circuit will require an LED logic level indicator at each Q output, a set of four logic level switches for the four DATA IN lines and a switch for the logic level on the LOAD line. Once the data word has been loaded and the operation fully understood, the shifting operation can be shown by pulsing the clock line with a de-bounced switch.

The Ring Counter

This is an interesting example of a shift register apparently masquerading as a counter, if the title is to be taken literally! It certainly can be used as a particular type of counter but it has a more interesting and useful application. It is frequently used for generating a 'walking one' pattern for scanning keyboards, multiplexing seven-segment displays and generating dot-matrix characters on video displays. It works very simply as follows.

The diagram of Figure 7 shows that the Q output of the last flip-flop is connected back to the D input of the first flip-flop, forming the 'ring' of the title. Thus, there is no actual input apart from the clock. To set it up it is necessary, when it is first switched on, to clear all stages (using a RESET line) and then SET just one of the Q outputs, say the first, to Logic 1. The 'preset' pin is used for this. If it is now clocked the 'Logic 1' shifts to the next stage to the right, its place being filled by a Logic 0. Every clock pulse performs the same action so that continuous clocking causes the Logic 1 to keep circulating. Naturally, when it reaches the Q output of the last flip-flop it will pass around the ring to the D input of the first, to keep the process going.

The sequence, for a four-stage ring counter, is 1000, 0100, 0010, 0001, 1000, etc.

Try this circuit out by wiring it up with a pair of 7474 ICs. Connect LED logic level indicators to all Q outputs and a low-frequency clock to the clock input. At about 1Hz it is possible to sit back and watch the 'one' circulate with ease.

The Twisted Ring Counter

This simple derivative of the ring counter just discussed is formed merely by taking the feedback from the not-Q output of the last flip-flop rather than the Q output (Figure 8). It is initialised in the same way – cleared and then a single 'one' put in. What happens when it is successively clocked? Would you be surprised to find out that the sequence is as follows?

1000, 1100, 1110, 1111, 0111, 0011, 0001, 0000, 1000, etc.

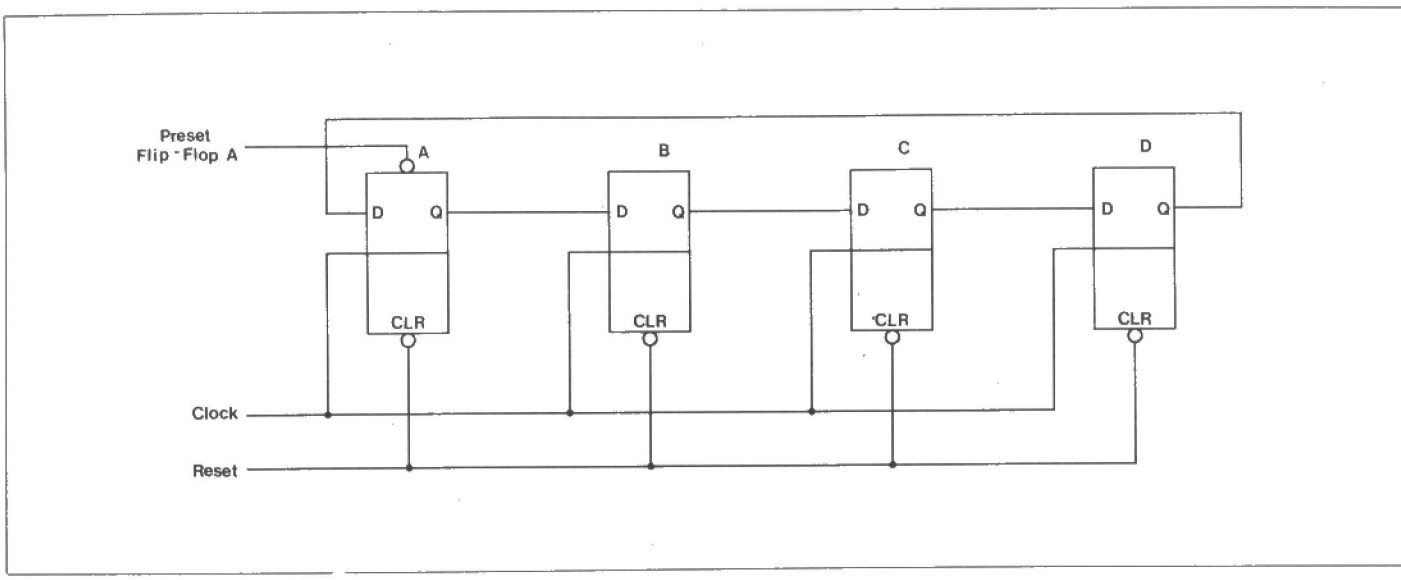


Figure 7. An example of a register with 'feedback', the Ring Counter.

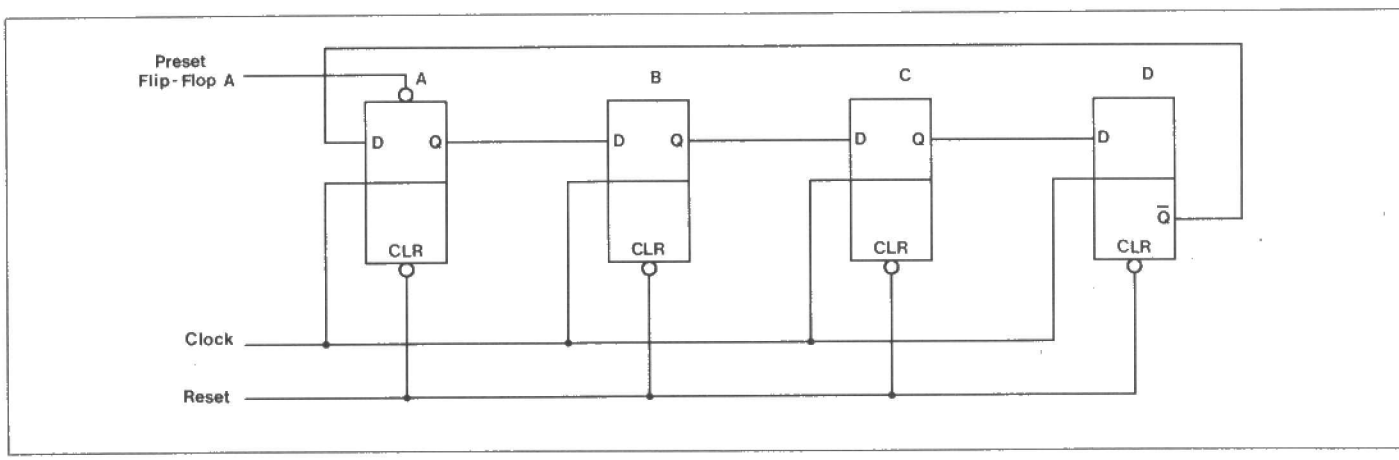


Figure 8. If the feedback is taken from the not Q output of the last flip-flop instead, the circuit of Figure 7 becomes a 'twisted ring' counter, with quite a different sequence.

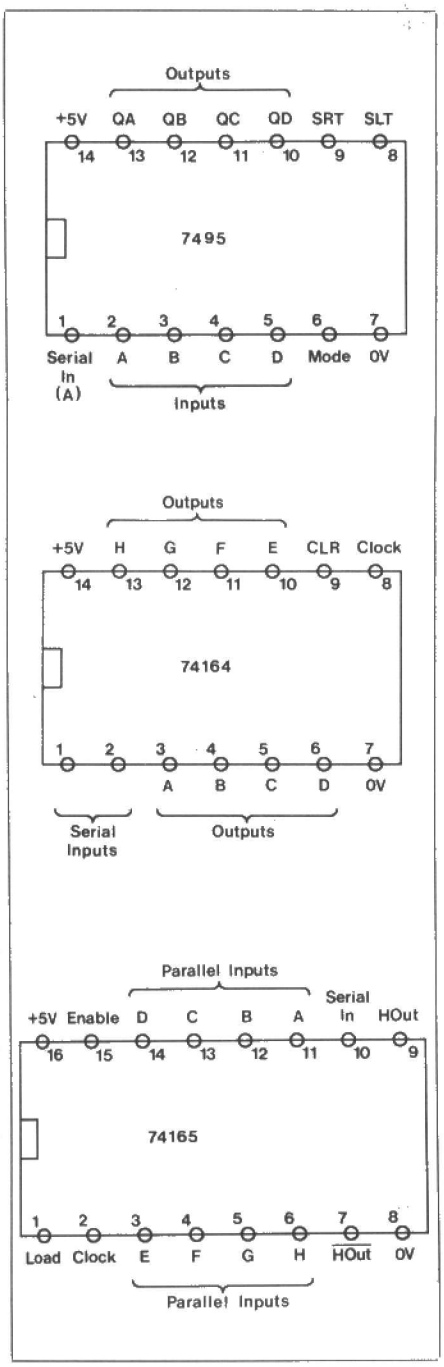

Outputs
+5V QA QB QC QD SRT SLT
14 13 12 11 10 9 8
7495
1 2 3 4 5 6 7
Serial In (A) A B C D Mode 0V
Inputs
Outputs
+5V H G F E CLR Clock
14 13 12 11 10 9 8
74164
1 2 3 4 5 6 7
A B C D 0V
Serial Inputs
Outputs
Parallel Inputs
Serial In
+5V Enable D C B A HOut
16 15 14 13 12 11 10 9
74165
1 2 3 4 5 6 7 8
Load Clock E F G H HOut 0V
Parallel Inputs

Figure 9. Pinout diagrams for some MSI shift registers.

It is interesting to notice that 'four' flip-flops give rise to 'eight' quite distinct states. In fact the number of states is always twice the number of flip-flops. Therefore, if five flip-flops are used, the number of states are ten – in other words we have a 'decade counter'. The fact that it is not the normal binary sequence for 0-9 is not really important as long as the state of the counter at any instant can be recognised by suitable decoding logic. Thus, if 0000 represents '0', then 1000 represents '1', 1100 represents '2' and so on. This code is called 'Johnson Code' and the twisted ring counter is often known as a 'Johnson Counter'. It has the characteristic of being a very fast synchronous counter, using shift register principles.

Further Work with Registers

The registers discussed so far have all been assembled from flip-flops and gates in individual IC packages. However there are, of course, fully operational registers contained within single packages. These often offer particular facilities that would be rather tedious to provide by interconnecting flip-flops and gates. A few of these will be discussed now, to round off this feature.

The 7495 Four-Bit Left-Right PIPO Register

This comes in a 14-pin DIL package and the pin-out diagram, together with those for the 74164 and 74165 registers, is shown in Figure 9. It is used as follows.

In this register all D inputs and Q outputs are individually accessible. It is possible to parallel load data into it and then either access it immediately (PIPO mode) or shift it left or right. Reversible registers normally require a fair amount of extra logic so having it 'all on one chip' is a decided advantage. It is possible to cascade these registers to make, for example, an eight-bit register.

There are two modes, known as SHIFT and LOAD. To go into SHIFT mode, the 'mode' input must be taken low. The negative edge of a clock pulse on SRT shifts data one stage 'to the right'. Similarly a negative edge on the other clock pulse, SLT, will shift the data one stage left. Thus, two clocks are needed, though in practice it could be the same clock switched between the SRT and SLT pins. Shifting works exactly as described for previous registers.

To go into LOAD mode, the 'mode' input is taken high and the data presented at the inputs LA – LD is entered into the register when a negative-going transition is made at the shift-left input (SLT). From this it can be seen that the SLT input is dual-purpose, shifting data left in SHIFT mode and loading data in LOAD mode.

The 74164 Eight-Bit SIPO Shift Register

This register, which has a right-shifting ability only, can handle serial-in data in the usual way, but can be pre-loaded with a Logic 0 or Logic 1 input as required. The data may be output in either parallel form or serial form.

For normal operation, one of the Serial Inputs is held high and the other becomes the data input. The Clear pin is also held high (taking it low momentarily clears the register). Data is shifted right one stage on every positive clock edge. To enter a Logic 1 into the register both Serial Inputs must be taken high. Taking either Serial Input low enters a Logic 0. It is essential that the clock is bounceless and noise-free.

The 74165 Eight-Bit PISO Register

This register has the versatility of being able to function as a straightforward Serial In Serial Out shift register or can be loaded with parallel data, which can then be shifted out serially.

For normal operation, the Enable pin is held low and the Load pin is held high. Data in is applied to the Serial In terminal and shifts right one stage on every positive clock edge.

To parallel load the register, the Load pin is momentarily taken low while the data input is applied to the parallel input pins A – H.

Shifting may be disabled at any time by taking the Enable pin high.

With the previous experience of connecting up and testing the other registers mentioned earlier, it shouldn't be too difficult now to devise schemes for testing the above three registers. One learns best by trying out a variety of circuits in this way.

An Oscilloscope Probe

There is sometimes a problem when using an oscilloscope to view the logic waveforms of TTL circuits. The input capacitance of the CRO may 'kill' the waveform at the test point and what is needed is some means of isolating the test circuit from these unwanted effects. This can be achieved quite easily using just two components – a resistor and a capacitor. The resistor is chosen to give attenuation together with the input impedance of the CRO, 10:1 being usual. See Figure 10. The capacitor is wired in parallel with the resistor R1 and is chosen so that the time constant of R1 and C1 equals the time constant of R_{in} and C_{in} for the CRO. This gives the widest possible bandwidth.

To take an example, the typical input impedance of a CRO consists of a parallel combination of a 1M resistance and a capacitance of the order of 10 – 50pF, say 30pF. Therefore, for 10:1 attenuation, R1 should be 9M. This gives the relation that:

$$9M \times C1 = 1M \times 30pF$$

$$\text{From which } C1 = \frac{(1 \times 30)}{9} = 3.3pF$$

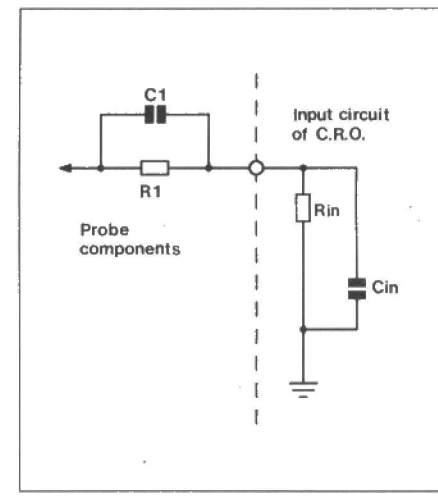


Figure 10. Probe design for a CRO.

It should be quite possible to assemble these two components into a discarded ball-pen case to make a useful probe, using a short length of co-axial cable between the probe and the Y input of the CRO.

'Hello Who's Calling?'

Bell's first telephone

by J.K. Hearfield Part 2

Central Battery Signalling

The earliest telephones were signalled by means of a 'magneto', or hand-cranked generator. Before making a call, the subscriber had to wind the magneto handle energetically several times in order to alert the switchboard operator, who then connected herself to the caller to find out what he wanted. This must have been rather a nuisance, but of course the subscriber had to do it or the call could never be placed. Once the call was finished, the subscriber was supposed to wind the handle again to let the operator know that she could clear the circuit. This procedure was known as 'ringing off', a phrase which survives long after its original meaning has disappeared. If the subscriber forgot, or didn't bother, there wasn't much the telephone company could do about it, and it must have been commonplace for circuits to be tied up long after calls had ended.

An obvious way to improve efficiency was to build into the switchboard some simple means of detecting when a caller finished speaking and hung up, and use this signal to alert the operator instead of relying on the caller to do it. Looking back from the vantage point of ninety years' steady improvement in technology, it seems perverse that the designers of the day chose to solve the problem in the way they did, but it should be remembered that relays were not always the cheap and reliable components they are today.

The solution chosen was to add what was effectively a third wire to the subscriber pair. Using the earth itself as one of the conductors had already been tried and found to be a bad idea, but using the earth as a third conducting path just for signalling was perfectly feasible, and continued to work well in one guise or another in the UK until at least the 1970s.

The magneto was itself not a low-cost item, and once it was no longer needed for 'ringing off', the way was open to remove it from telephones altogether, provided some equally straightforward way could be found for subscribers to initiate their calls. This proved remarkably easy. A telephone needs dc in order to work, and the earpiece is hung tidily on a hook when the telephone is not actually in use. So if the hook can be attached to a switch so that the circuit is broken when the telephone is 'on-hook' (another phrase from the distant past which inexplicably remains in common use, especially amongst telecommunications engineers) all the switchboard need do is monitor whether or not a dc path exists through the telephone. The current involved is quite small, because the power to energise the telephone's microphone is still drawn from the local battery, so the central battery need not be prohibitively large and expensive. This system was known as Central (or Common) Battery Signalling, or CBS. The logical next step of replacing thousands of small local batteries with one Central Battery (CB) was still however some way off, and will be discussed in the next article.

The Candlestick Telephone

Voltage detection at the exchange meant that the associated telephones could be smaller and 'portable' (within the limits of their cords). The classic candlestick telephone, typically as shown in Photo 1, exploited this freedom with a radically different shape. The transmitters in the first candlesticks were held in a rigid metal case, and users had the choice of either crouching over the instrument or picking it up (being careful to hold it vertical) in order to talk into the microphone. The user guide, which was engraved around the transmitter casing,

Photo 1. CBS table telephone.

Photo 2. Candlestick telephone, circa 1927.

often included a stern instruction to 'speak closely'. Later versions housed the transmitter in bakelite, an early brittle plastic, designed so that it could be swivelled to face the speaker. The weight of the earpiece operated the hookswitch, and the earpiece shape reflected that of the transmitter stand. The result was a functional and beautiful design which lasted for over fifty years.

The bellset was separate, and could be placed at the subscriber's (or his butler's) most convenient location. As well as housing the bell, it was a convenient termination point for cables from the outside world, the local battery, and of course the telephone itself. These bellsets, usually mounted high on a wall, survived for many years in houses where the telephone itself was modernised, and I was delighted to discover a beautiful example disguised by several layers of paint when I recently moved house.

The candlestick outlived the CBS system in the end, possibly because no-one could find a good reason for scrapping such a manifestly successful design. Candlesticks with dials became commonplace, see Photo 2, and if they were not as elegant as those without, they made up for it by providing a more efficient service.

CBS Telephone Systems

CBS telephones did not have dials, since all calls were still routed through the local manual exchange operator. The early CBS systems were intended for rural exchanges, and this restricted facility was all that was available for many years to some subscribers. And like the colour choice of the early Ford cars, Post Office subscribers could have any colour as long as it was black, though subscribers to the National Telephone Company could choose a nickel plated version.

The normal equipment consisted of a Telephone No.4 and a Bell Set No.5, and the standard CBS No.1 telephone installation is illustrated in Figure 1. The circuit of the telephone itself is shown in Figure 2 as it would have appeared at the time and redrawn with modern symbols. It differed from earlier telephones chiefly in its use of a lower impedance bell, connected not across the A and B wires but from the A wire to earth.

An earth connection on the A wire meant 'clear', because the earth could only be there when the telephone was on-hook. To make a call, the subscriber lifted the receiver earpiece from the hookswitch and the subsequent loop current was sensed at the exchange. In-coming calls were signalled when the exchange applied a large alternating voltage between the A wire and earth, causing the bell to ring. When the subscriber lifted the earpiece, the hookswitch was released, the ringing current path was broken, and the circuit back to the exchange was completed via the A and B wires.

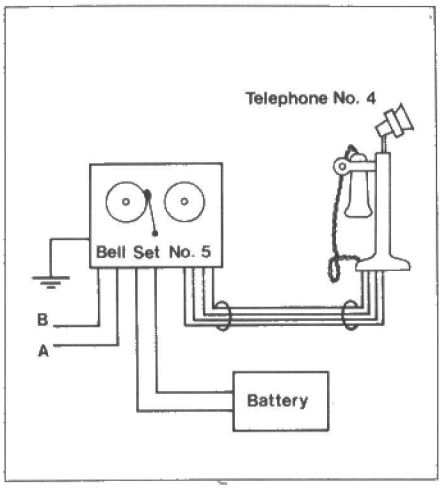


Figure 1. A standard CBS No.1 telephone installation.

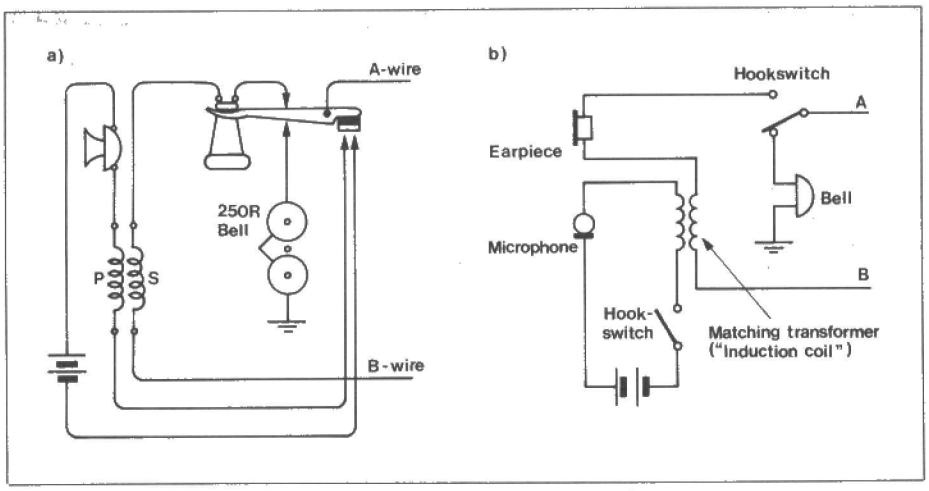


Figure 2. Principle of the CBS No.1 telephone, shown in both contemporary and modern styles.

There were disadvantages to this system. The bell would never ring if the A and B wires were acccidentally reversed anywhere along the route from the exchange to the subscriber's premises (a mistake it was all too easy to make). More importantly, the reliability of the system was governed to an uncomfortable degree by the quality of the earth resistance at the subscriber's premises. Telephone engineers had to explain to disbelieving subscribers that it was vital to water their earth connections in dry weather!

Photo 3 shows another early CB telephone. This elegant instrument not only incorporated its own bell, making a separate bellset unnecessary, but also provided an additional earpiece (for people with two ears, perhaps). The hookswitch design was less impressive, relying as it did on a long leaf-spring inside the handset cradle base which in practice was temperamental and difficult to keep in adjustment.

The simplicity and low cost of the CBS No.1 system ensured its survival for almost half a century, long after fully automatic CB systems had come to be taken for granted by most telephone users. Its disadvantages centred mainly around its use of an earth return path for signalling, and the CBS No.2 system attempted to get round these by introducing a blocking capacitor to separate the ac ringing path from the dc speech path. Figure 3 illustrates the essential difference between the two systems. In CBS No.2, ringing current flows through the 2μF capacitor and the bell, whilst dc current flows through the speech circuit. It was necessary to increase the resistance of the bell, which is effectively in parallel with the speech network, but this was easily done by wiring the 500Ω bell coils in series rather than in parallel. Capacitors cost much less by 1940 than they had done in 1900, so the economic effect of the change was not excessive, and CBS No.2 had the great merit of still not requiring a large and costly central battery.

Pay-Phones

Pay-phones for public use were available from an early date. In the 1880s

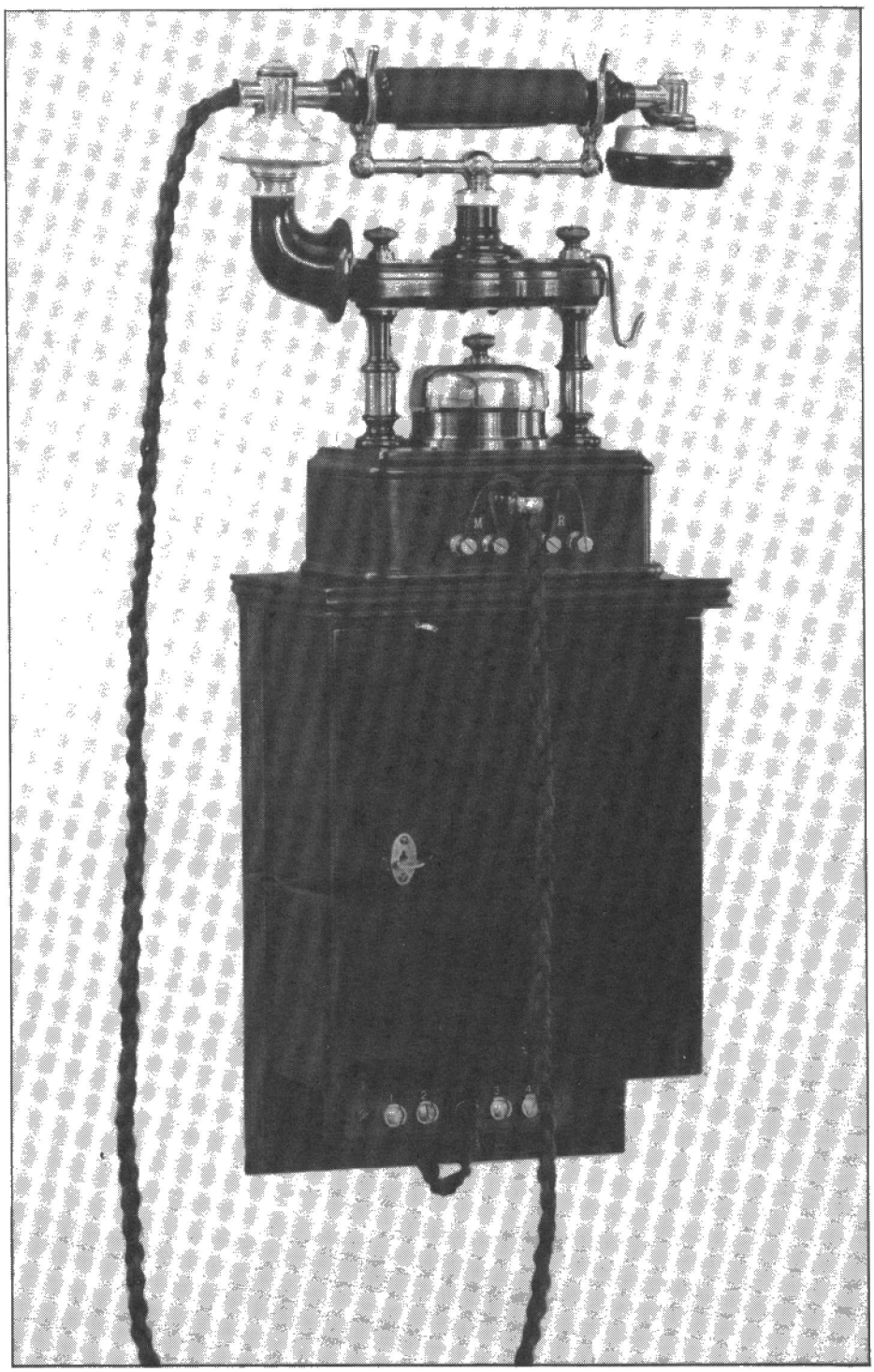


Photo 3. CBS telephone No.26.

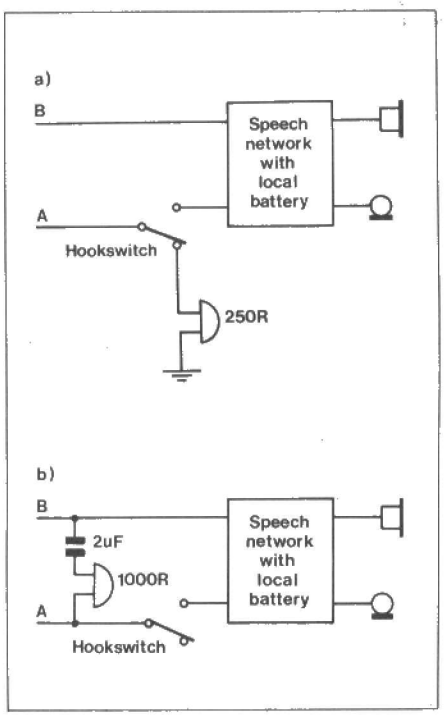


Figure 3. CBS No.2 replaced CBS No.1 about 40 years ago.

there were about twelve thousand private telephones in use, mainly by businesses, and in 1884 the Postmaster General authorised the telephone companies to create public call offices. The early pay-phones were installed in shops, post offices and railway stations, but telephone kiosks arrived in the streets as early as 1900. Some had coin-slots in the door, others had an attendant to collect the money, but whichever method was used it was possible for anyone with 2d to make a telephone call. Such permanent additions to the street also changed its appearance, sometimes to the dismay of the local council. After a number of unsuitable, grotesque or just plain ugly call-boxes had been installed by the various telephone companies, a design competition resulted in Sir Giles Gilbert Scott's classic red telephone box being introduced in 1927.

Style and Applications

Although most people will be familiar with the old types of domestic telephone many different designs were produced for different applications. House telephones and office intercoms used internal wiring to call instruments in another part of the building and were therefore cheaper to use than telephones connected to the public exchange. Railway companies used special telephones which needed little maintainance and could work cheaply over long distances. For many years the railways' private network was as large and complex as the public network. Gasworks and mines used special spark-proof telephones.

With greater use of the telephone, style and fashion became more important. As well as the nickel-plated instrument already mentioned, other inducements were used to persuade the subscriber. One Post Office telephone, similar in style to the Type 26, used

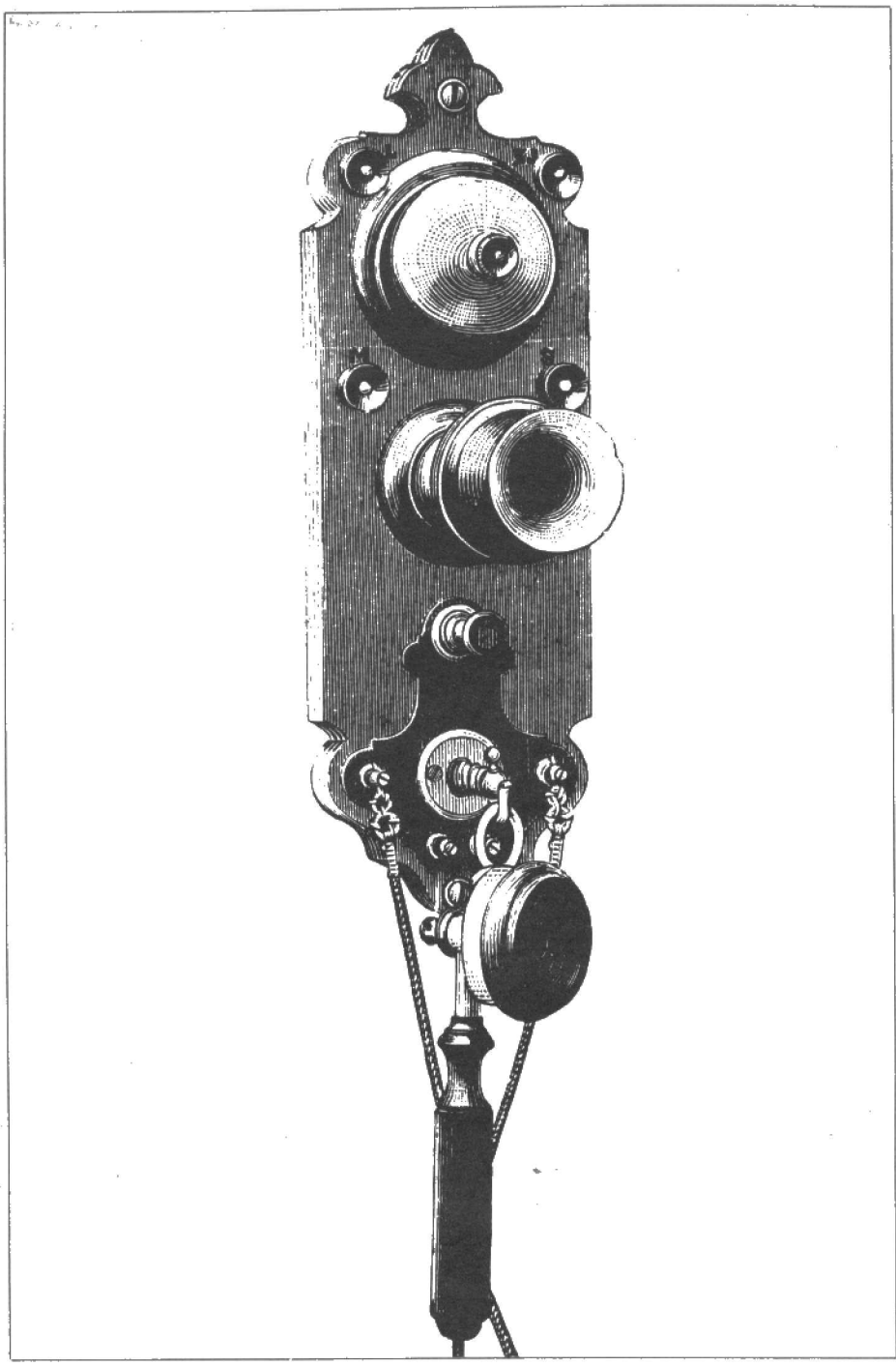


Photo 4. Combined servants call bell and telephone.

stained and varnished wood where possible and printed a woodgrain pattern on the remainder of the metal case. Another early telephone used small cow-gongs instead of bells to summon the subscriber. Some prosaic black telephones were made more up-market by the use of gold filigree transfers. If a household had to have one of these new-fangled instruments, then some effort was made to allow it to be as discreet, or as brash, as the subscriber wanted. In large houses where bellpushes were used to call the staff – see Photo 4 – it was sometimes possible to use the existing wiring to convert the system. 'Ringing for the cook' meant that it was possible to talk to the cook in her kitchen, instead of discussing menus in person, and the subscriber could call for the car without the need for the chauffeur to walk half a mile from the garage to the study.

Ownership of a telephone was however still rare. The industrialisation which followed the first World War created a much greater demand, which was satisfied by the more reliable mass-produced telephones to be discussed in the next article.

Acknowledgement: All telephone illustrations are reproduced by courtesy of the archivist at The Telecom Technology Showcase.

Further reading: Old Telephones by Andrew Emmerson, published by Shire Publications Ltd., Cromwell House, Church Street, Princes Risborough, Aylesbury, Bucks HP17 9AJ. Cover prize £1.25.

2 WAY 16 LITRE LOUDSPEAKER SYSTEM

by Dave Goodman

Introduction

Optimised cabinet designs can be regularly found in the plethora of electronics books and magazines available, and doubtless all have their relative strengths and weaknesses according to latest theories and materials available! We all perceive sound in different ways according to needs, age, personal taste, surroundings and many other variable factors which shape – or cloud – our judgement of what sounds good or bad; therefore, no claims of superiority and geometrical excellence or latest technological marvels are made for this design which is a small 2 way loudspeaker cabinet measuring 25cm x 21cm x 41cm, which is equally suitable for use in home hi-fi and studio or stage monitoring applications. The prototype frequency response, shown in Figure 1, extends from 40Hz to 18kHz with power handling capability up to 40 watts*. Just what the design has to offer however, is:

A. Ease of construction due to using simple chipboard panels for the box; building the prototype took approximately 2 hours – including sawing and gluing the wood!

B. Front baffle supplied drilled and pre-cut to take loudspeakers and port tube supplied in the kit.

C. Very high quality bass/mid range driver with treated fibre glass cone which produces a smooth, clearly defined presence to the sound.

D. SOA tweeter, active horn loaded and formerless ferro-fluid cooled, offering good transient response and spatial dispersion.

The cabinet dimensions were calculated from T/S data available for the driver and previously published in the March/May 1987 edition of Electronics.

Figures 2 and 3 show the manufacturers response curves for the two speakers both on axis and 30 degrees off axis. The smooth bass response of the driver can be seen from the gradual low frequency roll-off between 100Hz and 20Hz in Figure 2, whereas the slope characteristic changes completely once the driver is fitted into the cabinet as seen in Figure 1.

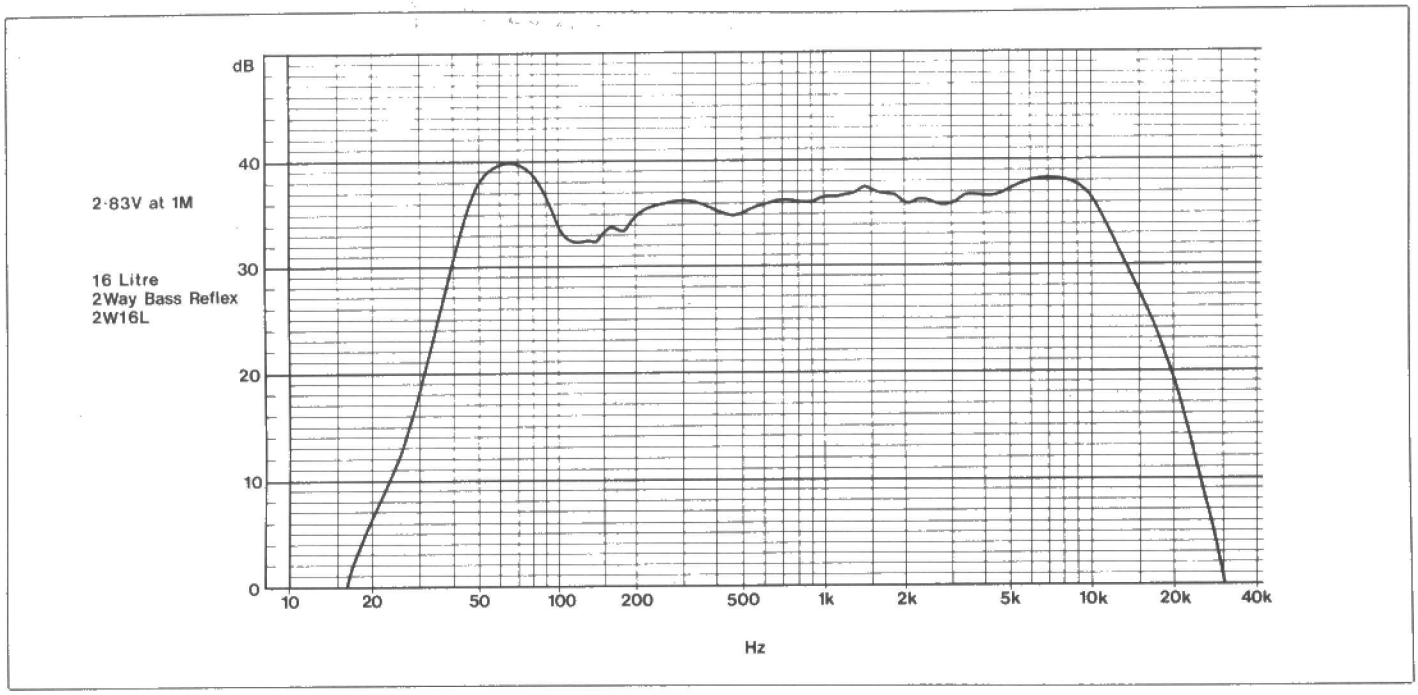


Figure 1. Total system frequency response.

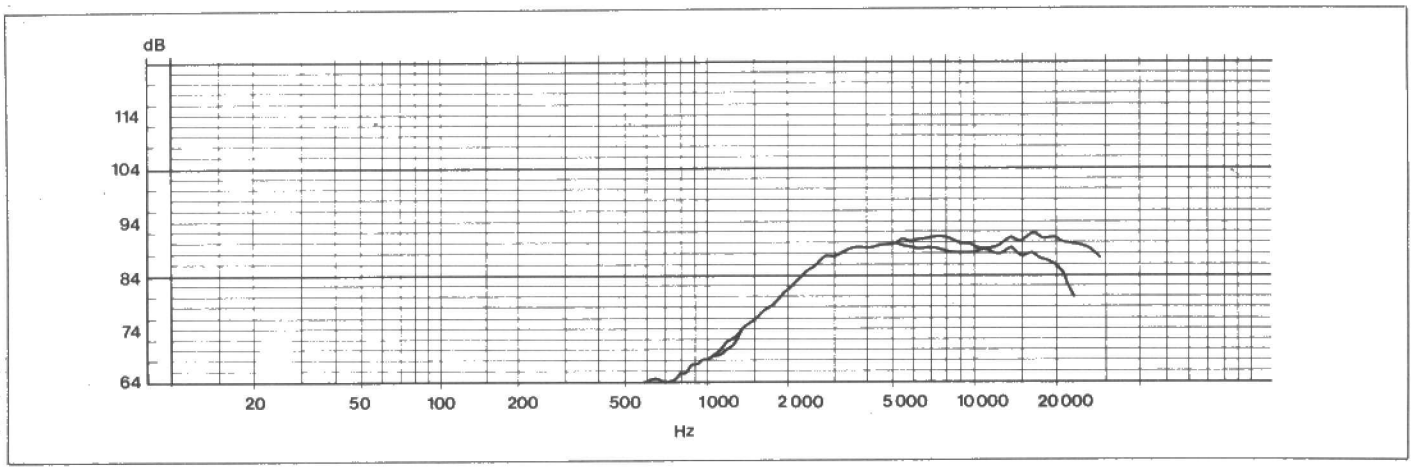


Figure 2. Bass speaker response curve.

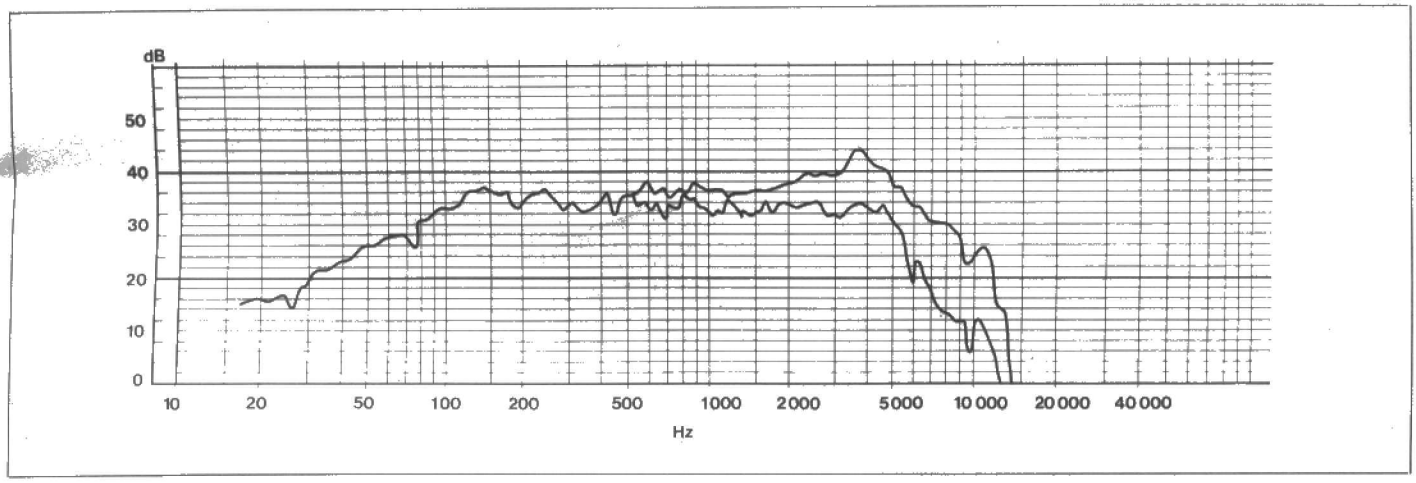


Figure 3. HF tweeter response curve.

Cabinet Construction

Five panels are required for the cabinet as they are *not* supplied in the kit. The cheapest and possibly easiest material to use is 15mm thick, High Density Chipboard of the type used for flooring. Melamine and veneered chipboard is suitable, if the laminate is removed from butt joints before gluing, or marine plywood can be used. Chipboard is very cheap and can be veneered, painted or wallpapered to taste quite easily.

When cutting panels, particular care should be taken to ensure straight and square edges are formed. At the least, large air gaps will appear along the joints, at worst – the finished cabinet will not fit together properly. Most large wood yards have power saw facilities and will cut these panels very accurately for you.

Please note that the dimensions shown in Table 1 are based on 15mm thickness material. Any deviation from this thickness will require an alteration to

Cutting List

Material 15mm thick HD chipboard

1 panel	410 x 180mm	(1)
2 panels	410 x 250mm	(2) & (3)
2 panels	250 x 210mm	(4) & (5)

Material 2" x 1" (50mm x 25mm) prepared

2 horizontals 138mm (6) & (7)

2 verticals 410mm (8) & (9)

Also required are:

18 csk chipboard screws 38mm

6 wire nails 30mm

White wood resin 'W' adhesive

Contact adhesive or glue gun

Table 1. Cutting list.

the 210mm width measurement on panels (4) and (5) only, to allow for the change. Figure 4 gives the panel layout.

Suggested Assembly

If reference is made to Figure 5 panel assembly drawing, it can be seen that panels (2) and (3) require three clearance holes drilled along each of the long edges, for the chipboard screws. Make the holes approximately 7mm in from the edges and counterdrill them to take the screw heads. Note that the larger (68mm) of the two small holes cut out of the front baffle is positioned downward towards the base! Top and base panels (5) and (6) also require 4 clearance holes to be drilled, one positioned centrally along each side and 7mm in from the edge, and these too should be counter drilled as before.

Spread a liberal amount of resin adhesive along the front inside edge of panel (2) and offer it up to the left hand edge of the baffle; insert three screws as shown and tighten them up, ensuring both panels are aligned precisely at the front and ends. Repeat the same procedure as before for the right hand panel (3) and baffle.

Next, fit the back panel (1) temporarily in place using screws only (no glue!) and assemble top and base panels (4) & (5). Apply adhesive to the top edge of panels (2), (3) and baffle only, position panel (4) and insert the four screws. Tighten each screw and with a clean rag, remove any excess glue that may have squeezed out from the joints. Repeat the procedure for base panel (5) as before. Finally, remove all six back panel screws and take out panel (1); you may have to knock the back panel out by inserting a hammer or block of wood through the front speaker cut out and apply a hefty wack!

Framework

A simple framework comprising 4 lengths of 2" x 1", or metric equivalent, is required for bracing the back panel (1). Earlier assembly details suggested that the back panel could be held in place with six screws inserted through side panels (2) & (3). If intending to cover the cabinet after construction and also to allow for the

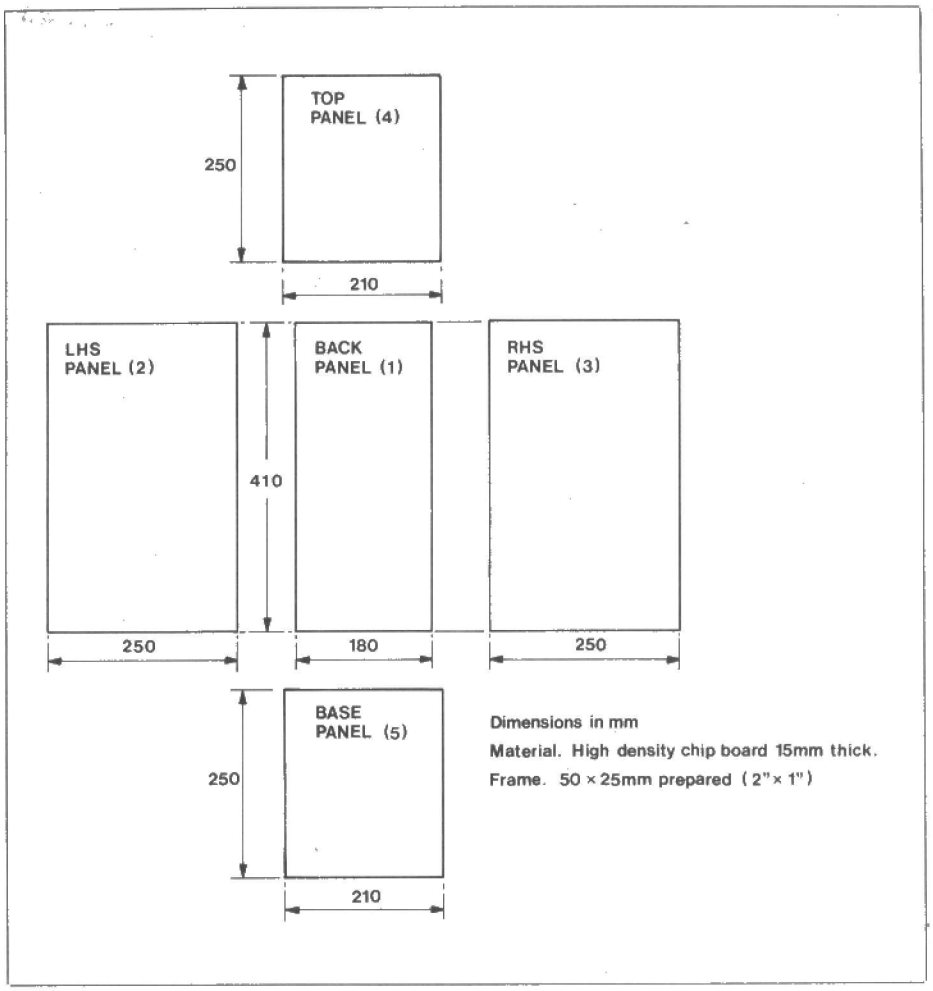


Figure 4. Panel cutting details.

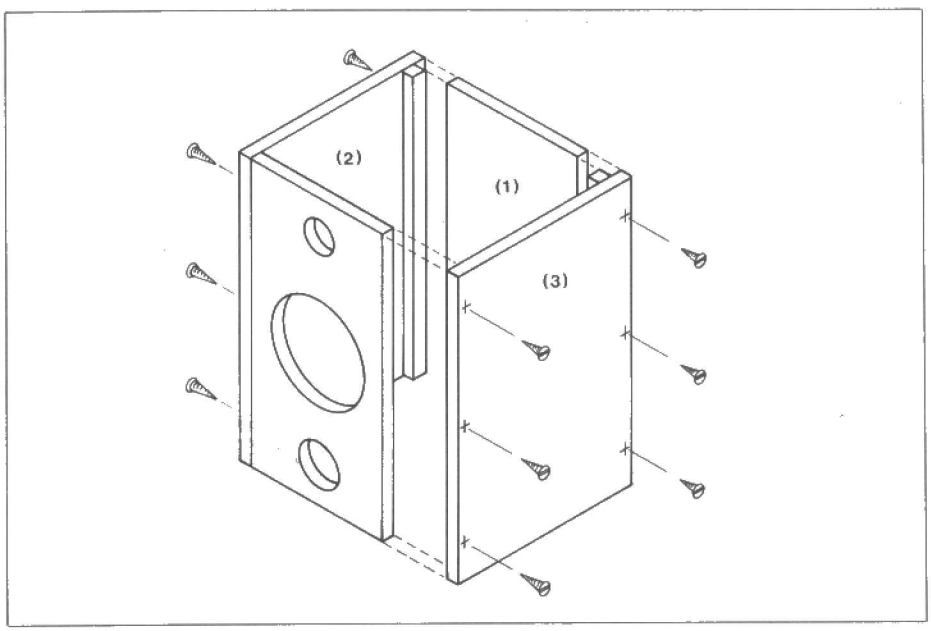


Figure 5. Panel assembly.

back panel to be removable, or perhaps you would not want to damage any external finish on the side panels, then the screws can be inserted through the back panel and into the frame instead.

Figure 6 shows details of the framework assembly and the inset drawing emphasises a 15mm recess to allow for the thickness of the back panel. Both vertical braces (8) & (9) are cut to the same length as the side panels (410mm), whereas the two horizontal brace (6) & (7) lengths must allow for the particular thickness of material in use. Prepared timber can vary in size, but is usually 21mm for 1 inch, hence the length of 138mm being suggested in the cutting list! Glue the framework in position and use panel pins or wire nails to secure each timber while the assembly dries out and re-check the accuracy of the 15mm recess measurements.

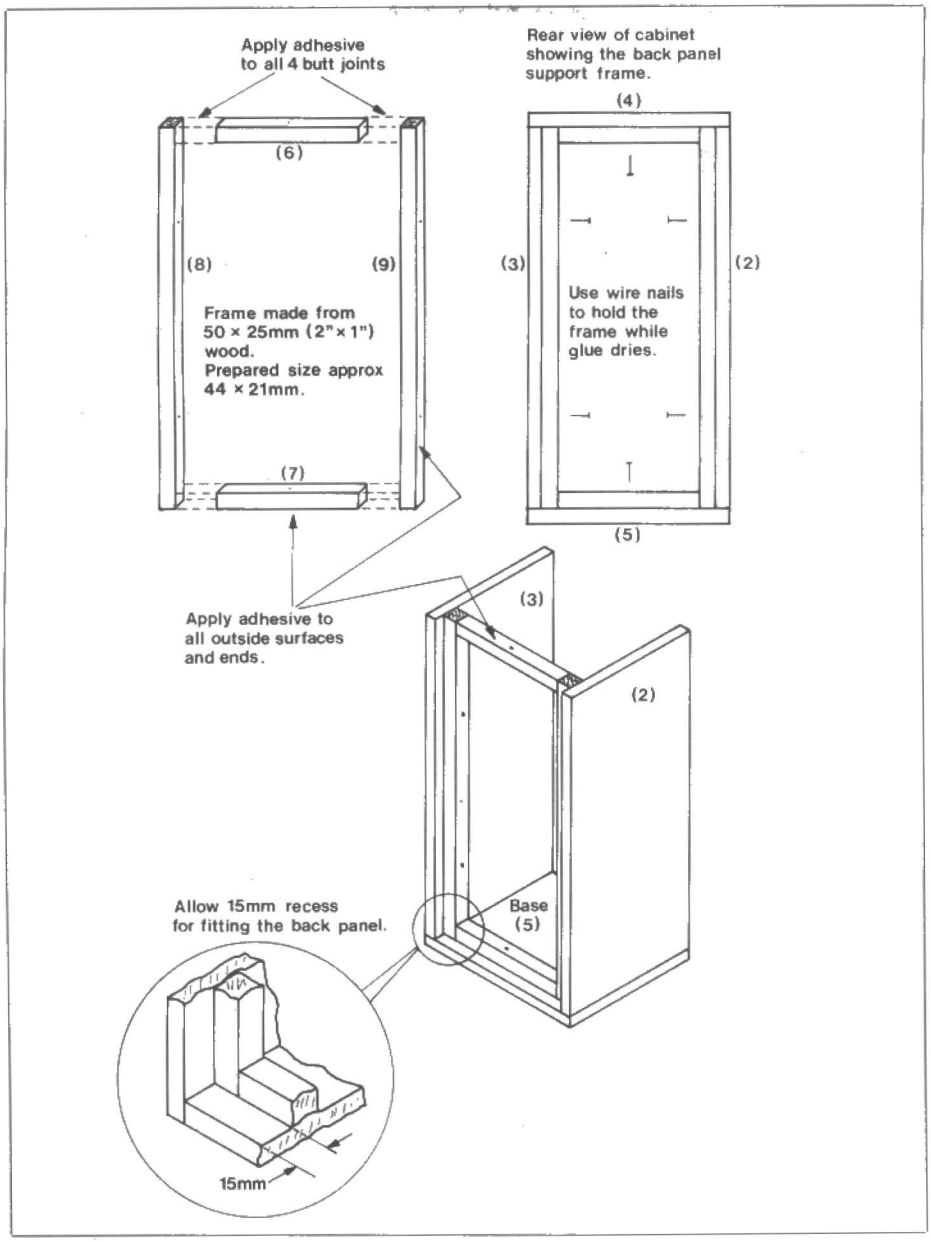

Apply adhesive to all 4 butt joints
Rear view of cabinet showing the back panel support frame.
(4)
(6)
(8)
(9)
(3)
(2)
Frame made from 50 × 25mm (2" × 1") wood. Prepared size approx 44 × 21mm.
Use wire nails to hold the frame while glue dries.
(7)
(5)
Apply adhesive to all outside surfaces and ends.
(3)
(2)
Allow 15mm recess for fitting the back panel.
Base (5)
15mm

Figure 6. Back panel framework.

Port Tube

A length of plastic tubing is supplied with the kit as a resonator for tuning the cabinet. This 'port', as it is known, relies on the effect of Helmholtz resonance for greater LF acoustic output and lower cut off frequency than would be possible in a sealed box version. The tube is simply inserted into the 68mm cut-out, which should be a tight fit, and secured in place with a suitable adhesive or glue-gun. Apply the adhesive from the inside as shown in Figure 7.

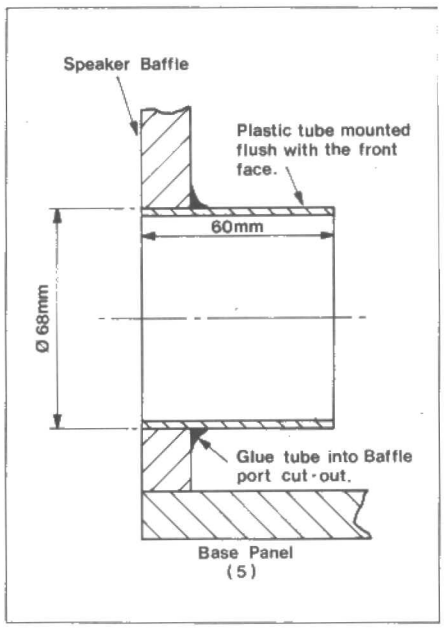


Figure 7. Port tube fitting.

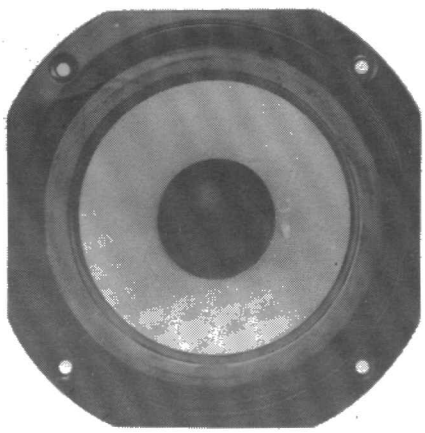
Wadding

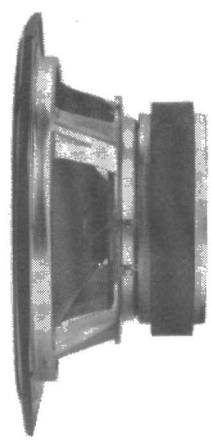
Use a sharp knife or scissors to cut out five sections of acoustic wadding. Each piece is glued onto the inside surface of

Specification

Fibre Glass Driver – YN25C

Chassis	173 x 173mm
Baffle	146mm
Fixing	124 x 124mm. 5mm clear
Flux Density	13000 Gauss
Frequency Response	35Hz to 5kHz
Power Handling	40 Watts *
NOM Impedance	8 Ohm
Sensitivity	91dB (1W at 1m)
Coil Diameter	25mm
Free Air Resonance	40Hz
Moving Mass	0.0114kg
Mechanical Q	1.01
Electrical Q	0.22
Total Q	0.18 **
Equivalent Air Load	30 Litres





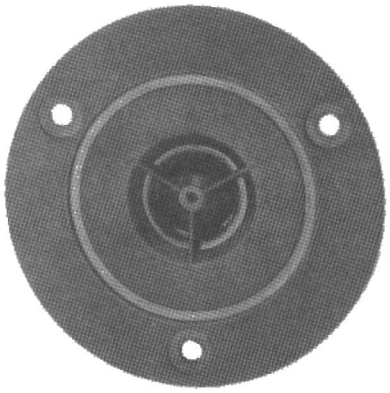
* Power specification to DIN 45573. For further reading on this matter, see Electronics, March to May 1988, "Loudspeaker and Amplifier Specifications".

** Tests on the prototype have shown the total Q to be double that specified i.e. Total Q = 0.36. Although this figure may appear insignificant, its relevance in cabinet design is extremely important and this 16 Litre design is based on the revised figure.

Specification

Miniature Dome Tweeter – FD94C

Chassis	74mm
Baffle	48mm
Fixing	3 x 4.2mm
Flux Density	10600 Gauss
Frequency Response	2500Hz to >25kHz
Power Handling	40W @ 5kHz *
NOM Impedance	8 Ohm
Sensitivity	91.3dB (1W at 1m)
Coil Diameter	10.5mm
Free Air Resonance	2900Hz ± 400Hz



the top, base, side and back panels, but NOT onto the front baffle! Ensure that the piece glued onto the back panel does not foul the framework with the panel in place as this will prevent correct fitting. Also, drill a small hole in the back panel at least 30mm in from one edge and which is just large enough to allow the speaker connecting cable to pass through.

Speakers

Before installing the speakers into the cabinet, check that you can identify the positive (+) terminals on both units. It is important to wire speakers so that they have the same phase relationship, especially when two or more are used in stereo or quad applications. Here, the phase relationship just requires each speaker cone in the system to move in the same direction. If one speaker cone were to move inward while the other cone moved outward, then frequency cancellation takes place; this effect becomes prevalent at low frequencies and results in loss of bass. The tweeter + symbol is stamped into the plastic body moulding and is easy to spot, but the YN25 driver may only have a red dot painted on the chassis, close to the terminal block. If in doubt, connect a 1.5V battery to the speaker terminals so that the cone moves. If the cone moves inward to the speaker, reverse the battery connections so that the cone now moves outward and away from the speaker: the terminal connected to battery positive is now the speaker positive (+) and should be marked for later reference.

Now solder a length of black wire onto the tweeter negative terminal and a length of red wire onto the tweeter positive terminal. Place both wires through the small baffle cut-out, into the cabinet and out again through the driver hole. Insert the tweeter into the baffle from the outside, as shown in Figure 8 and secure in position with 3 x No 6 self tapping screws. It will be found helpful if small pilot holes are first made in the baffle before fitting the screws.

Solder any one end of the 1μF bi-polarised capacitor to the driver speaker positive terminal and the remaining end to the red tweeter wire, keeping the capacitor wires as short as possible to prevent them from shorting out. Solder the remaining black tweeter wire to the driver

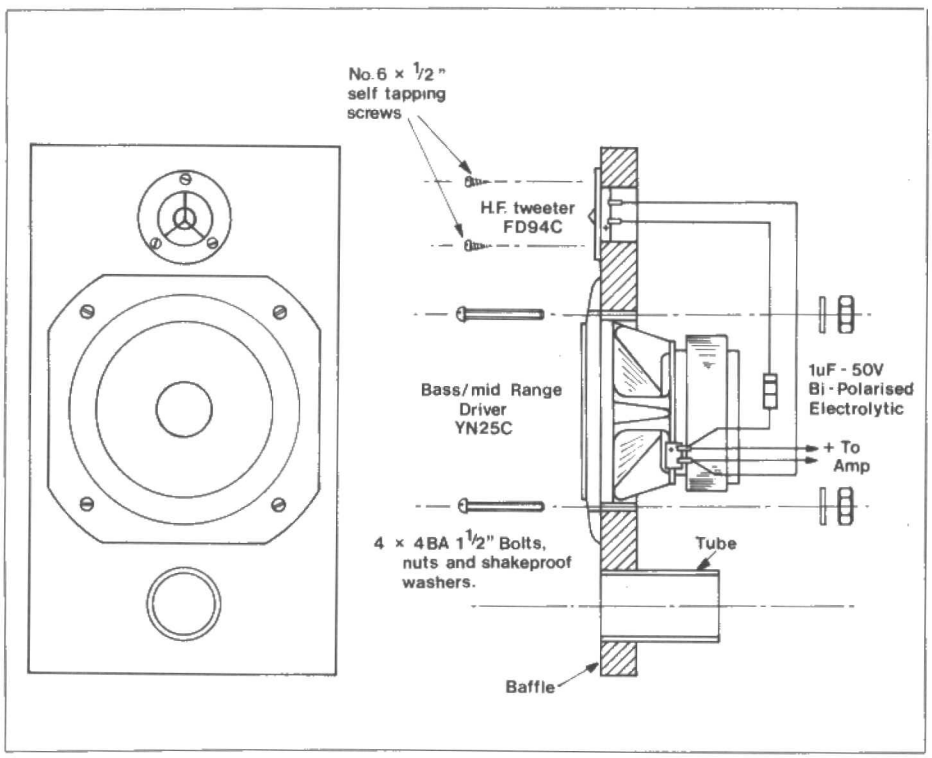

No.6 x 1/2" self tapping screws
H.F. tweeter FD94C
Bass/mid Range Driver YN25C
1uF - 50V Bi-Polarised Electrolytic
+ To Amp
4 x 4BA 1 1/2" Bolts, nuts and shakeproof washers.
Tube
Baffle

Figure 8. Fitting speakers.

negative terminal. At this time, it is convenient to terminate the cable or wires required for connection to the amplifier. This may be done directly or via a connector terminal as preferred. Whatever method is adopted, remember to identify the positive wire and solder it to the driver +V. Also, connect the negative or return common to the driver −V terminal as shown.

Mount the driver onto the baffle and align the four bolt holes, insert 4 x 4BA 1.5" bolts. Fit a shakeproof washer and nut onto each bolt, from the inside, and tighten each one accordingly. Placing a small blob of glue over these nuts will help prevent them from shaking loose in use and is recommended. Take care that adhesive is not spilt onto the speaker cone whilst doing this.

Final Assembly

Insert the connecting cable through the hole in the back panel and fit the panel in place. If you are using a connector terminal, solder the two speaker wires to each terminal and mount the block over the wire-hole; otherwise squirt some glue down the wire-hole to seal it off. Screw the panel firmly in place to complete the cabinet as shown in Figure 9. The order in which you carry out the various assemblies is not important and the details are offered as a guide. For instance, if intending to veneer or decorate the cabinet in some way, this should be done before mounting the speakers – for obvious reasons!

Conclusion

A few points to remember are:

(1) Handle loudspeakers carefully and protect them from contact with adhesive and damage from screwdrivers.

(2) Ensure polarity markings are found correct before wiring.

(3) In the cabinet, fill any gap along joints with a mixture of glue and sawdust to make air tight. Silicon rubber could also be used for this purpose, especially around the back panel and wire-hole.

(4) Do not subject the speakers to abuse by continuously over running them. Amplifiers rated 20 to 30 Watts RMS are generally suitable for use with this system.

(5) In a cabinet of this size, low frequency bass performance is very dependent on the size of a room and the position within the room, that is being used. Similarly, high frequency performance is dependent upon local furnishings and listening positions within the room and is, therefore, worthy of some consideration to obtain the best possible performance.

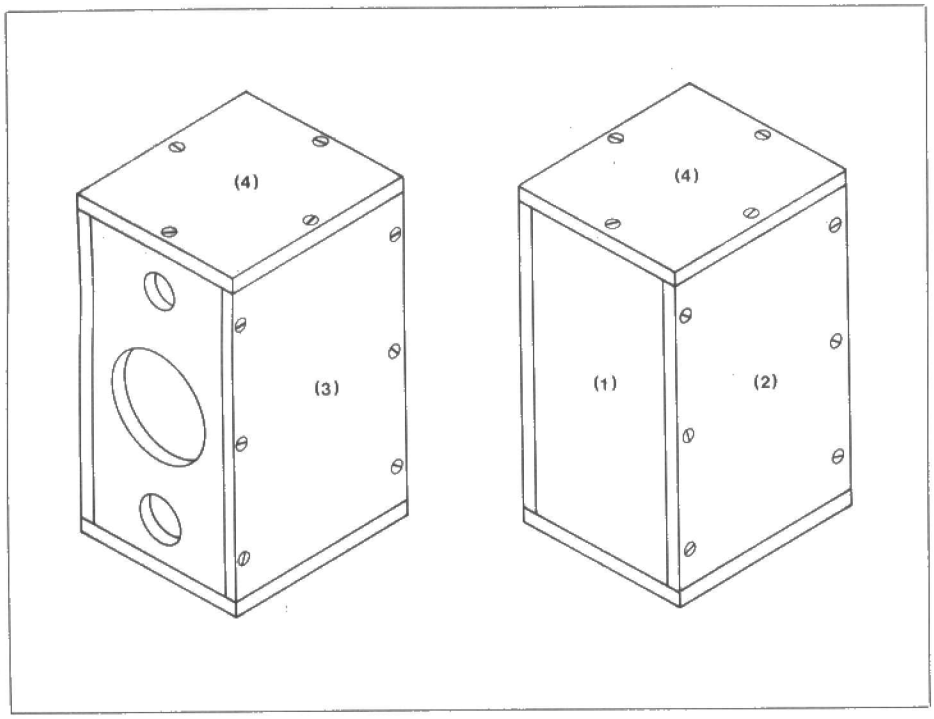


Figure 9. Final assembly.

2W16L CABINET PARTS LIST

Bass/Mid range Driver		(YN25C)
Miniature Dome Tweeter		(FD94C)
1μF Reversolytic Capacitor		(FB97F)
No.6 x 0.5in Slf-Tpr	3	(BF67X)
4 BA x 1.5 inch Bolt	4	(LR52G)
4 BA Nut	4	(BF17T)
4 BA Shakeproof	4	(BF25C)
Wire 3202 Red	1m	(XR36P)
Wire 3202 Black	1m	(XR32K)
Acoustic Wadding		(RY06C)
Port Tube 60		(YT03D)
2W16L Baffle		(XJ69A)
Optional		
Impact Adhesive		(FL43W)
Rubber Sealer		(YJ91Y)
Glue Gun		(YP71N)
Glue Sticks		(FS97F)
Lever Terminal 2-way		(BW72P)
HiFi Loudspeaker Cable		(XR72P)

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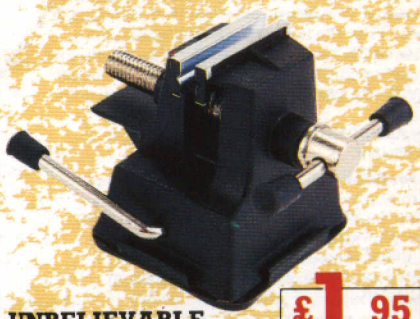

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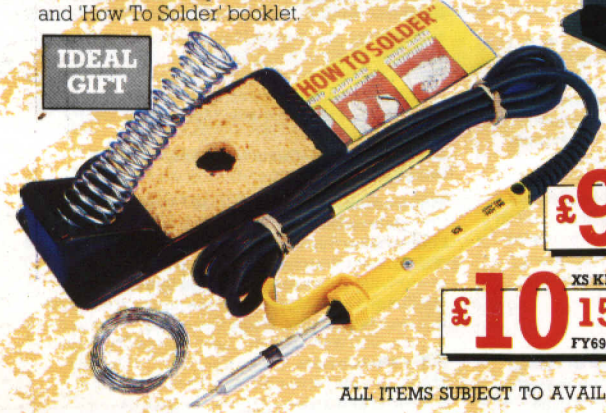

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